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STUDY AND INVESTIGATION OF SPECIALIZED ELECTRO-ACOUSTIC TRANSDUCERS FOR VOICE COMMUNICATION IN AIRCRAFT

Appendices 1 – 6

Contract AF33(616)-3710 – FINAL REPORT

Task No. 43050

February 1959

WESTERN ELECTRO-ACOUSTIC LABORATORY, INC.
LOS ANGELES, CALIFORNIA

Best Available Copy

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WESTERN ELECTRO-ACOUSTIC LABORATORY, INC.

Los Angeles

APPENDICIES

Text Outline

APPENDIX 1.0

SUMMARY OF CONFERENCES WITH PANEL OF EXPERTS

APPENDIX 1

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Appendix

1.0 Introduction

Upon completion of the Transducer Chart, Table II, a series of conferences was undertaken with the Panel of Experts to come upon a general agreement as to the possible combinations of transducer-coupler-shield which had promise of good performance, reduced size and weight, and improved comfort tolerance, and should be subjects of intensive investigation. Evaluation techniques were also intensively reviewed. The discussions leading to these conclusions, and the detailed design of the exploratory investigations are summarized in Appendix I, and represent a thorough and stimulating re-appraisal of voice communication possibilities and techniques for evaluation. In the course of the Panel conferences we were at first at pains to elicit individual thinking and suggestions. Thereafter the contributions were combined into the form presented in Section A-1, and sent without identification of source to the Panel for criticism. As presented in this Appendix, the source or sources of each suggestion are indicated by initials with each comment.

The key to the initials which identify the contributors by individual or organization are as follows:

HF - Harvey Fletcher

WS - William B. Snow

BBN - Bolt, Beranek and Newman - Leo Beranek, Francis Wiener

CH - Cyril Harris

GP - Gordon Peterson

WR - Wayne Rudmose

RCA - Radio Corp. of America - Willard Meeker, Ma[illegible] Touger

DM - Dan Martin

RB - Robert Benson

PV - Paul Veneklasen

HL - Haskins Laboratory - Alvin Liberman, Franklin Cooper, Katherine Harris

NEL - Navy Electronics Laboratory - Robert Gales, John Webster

GM - George Miller

JL - J. C. R. Licklider

GC - General Comment

1.1 Prospective Transducers - Microphones

1. The following transducers seem most feasible and therefore should be evaluated for relative usefulness:

a. Electrodynamic,
b. Ring Armature,
c. Electrostatic,
d. Piezoelectric. BBN, WBS, HF, RCA, WR

2. The absence of the carbon microphone was noted in our list of speech projectors. Perhaps familiarity has bred contempt. RCA, GC

3. The following types of microphone systems should be evaluated for relative performance: pressure, pressure gradient, throat, ear, head contact, each with and without a noise shield. GC

4. The addition of chest contact has been suggested. CH

5. The possibility of a microphone inside the mouth has been suggested. In most places where a pickup might be placed one might expect considerable alteration even of vowel sounds, by favoring one resonant region over another. The most likely place would be just inside the lower teeth since the edge of the lower teeth is effective in consonant production. A tiny cylindrical barium titanate transducer might be imbeded in a dental plate. GP, PV

6. A probe tube in the nose should be tried. GP, PV

7. Dan Martin's long experience with the throat microphone concludes that this device is hopeless for high intelligibility in noise. It can be better than marginal in the quiet. The failure is in consonant production. DM

8. The ear microphone has been investigated by many groups but it is doubtful whether the proper data has been taken for our purpose. RCA, PV, NEL

9. A study of the ear microphone at CID using an Air Force receiver in a doughnut cushion produced intelligibility equal to an open-air microphone in a noise field of 118 db. RB

10. All of our experts consider it extremely unlikely that a speech pickup can be successful unless it is close to the lips and teeth where the consonant sounds are formed. In other words, we can only lose information as we move away from the lips. Exploratory tests should quickly show what information is lost. GC

11. It is most important to evaluate the effectiveness of pressure vs. gradient microphones. Harvard EAL

experiences tends to discredit the gradient type.
BBN, PV, WR, WS, DM, RCA

12. A Baldwin project has compared pressure vs. gradient microphones in a mask. The results favor the gradient type. DM

13. Some noise cancellation should be expected from a directional microphone, in fact this may be one mechanism of the gradient type. WS

14. A pressure microphone may have some advantage over the gradient microphone because the single opening can be closer to the lips. WS, PV

15.. There is some evidence that a gradient microphone alters speech sounds, especially those involving nasal resonance. RCA, DM

16. Besides the pressure vs. gradient argument, another major factor has existed in microphone pickup, namely the obstacle effect which exists in both types. None of the microphones has been small enough to avoid direct impingement of the breath on the microphone with resulting alteration of speech sounds. Furthermore, the baffle between the pressure sensing openings detracts from pure differential action. The true differential effect might be approached by two small probe tubes. PV

17. The possibilities of a thermistor bead should be considered as a microphone because of small size. A pair of thermistors might be arranged as a gradient microphone. J. A. Becker and Howard Christenson are the experts at Bell Telephone Laboratories. HF

18. A "diversity pickup" microphone array has been suggested in which the optimum position regarding S/N would be used for each sound or each frequency range. JL, RB, WS

19. On the basis of known data a dual microphone is suggested for optimum noise exclusion. The idea is to pick up the consonants where you can get them best and to pick up the vowels where you get rid of the noise. For example, if the ear microphone proves to have the best S/N for low frequencies, and the lip position is essential for detection of consonants, and good noise shielding can be provided at high frequencies, then we might pick up the low frequencies at the ear and the high frequencies at the lips.

20. In testing new techniques such as head contact or the ear microphone, the question is how much time should be spent in assuring that we have the optimum

type of pickup? It is not necessary that the transducer be the best one for the purpose in terms of economics or adaptability; rather it is the method which is being tested and we must simply show that ~~the tra~~nsducer is not the limiting element in the method. GC

21. In regard to probe tubes, it has been questioned whether a termination which is appropriate at sea level will remain optimum at a high altitude. RCA

22. It may be possible to artificially accentuate cues for consonant sounds by producing auxiliary turbulance in the breath stream. The baffle of a close-talking microphone may perform this function. It may be that this form of distortion or alteration of speech sound is not unfavorable. GP

23. The eventual limit for communication in noise will be the maximum vocal output. It has been suggested that this limit may be overcome by using an artificial larynx, possibly including also a tube with an artificial airstream for the production of consonants. WS

24. We cannot locate data on the maximum sound pressure level which can be generated at a very close talking position. This data was taken at Harvard but apparently not recorded. It must be measured as soon as possible. WR, PV, BBN

25. One feature of the AIC/10 System most needs improvement: the microphone response is most unfortunate; the midrange dip occurs in the frequency region most important to intelligibility, and the high frequency peak gives the unpleasant quality and is not of any value for intelligibility. The dip tends to accentuate the peak. This is less so at high altitude. DM, WS, BBN, PV

26. Since the S/N achieved at the microphone is favorable except at high frequencies in the AIC/10 system, the boosting of the high end is wrong for the achievement of better intelligibility. Experience shows that given good S/N, overall equalization is not an important factor in intelligibility. In other words, equalization is useful only for achievement of better S/N which might well be the case at the listening end. GC

1.2 Prospective Transducers - Receivers

1. Among transducers for receiver units the same four types (see above) are suggested for evaluation. In both the piezo and electrostatic types humidity will be a problem. GC

2. The air modulated system has been suggested for a loudspeaker or for a receiver unit; is undergoing further study, sponsored by the Signal Corps. - Mr. Farella - in a contract with Stanford Research - Vince Salmon. BBN, WS

3. The ionophone or ionic source should be added to the electro-acoustic transducer list, for possible use as a loudspeaker only. The ionophone is being developed by the Du Kane Co. in St. Charles, Ill. RB, RCA

4. In the morphological chart, coupling by means of mechanical contact to the head had been omitted. This should be added. RCA

5. The Thermophone should be added to the transducer list. What is the frequency range? WS

6. For the sake of completeness, add to the list of electroacoustic transducers - the electrophonic phenomenon: electrodes are placed on head, attempting to directly excite cochlea. Reference: Davis - "Hearing and Deafness." Reports that pain threshold is reached only a few db above the aural threshold. RB

7. RCA is exploring the power limitations of their present earphones. The question in regard to our program is whether the units have enough power capacity to establish a useful signal-to-noise ratio outside an ear using an earplug, in other words, for external sound levels in excess of 130 db. Except for limited power handling capacity, there is potentially available communication using earplugs for an additional 15 to 20 db of noise. This possibility is of great importance for use by field mechanics around turbojet engines. PV

8. Increase in power capacity of the receivers will be part of RCA's product improvement contract and hence may be incorporated in our program by reference. RCA

9. In regard to the engineering analysis of other transducer types, the RCA group (Joe Hartley) has been making an analysis of the electrostatic type (in accordance with Hunt's book) which has not been reported. This may be made available to us. RCA

10. A liquid transmission system for coupling to the head has a major limit. The wave length of sound in the liquid is too long for a good radiation impedance around the edge of the coupling fluid, hence energy is by-passed. Mechanical coupling may be the best way. But then there is the skin compliance shunting the force, i.e. between force application and the skull. HF, PV

11. Silicone putty has been suggested instead of liquid coupling. PV, DM, WR

12. **What are the properties of Silicone putty? A liquid at low frequencies and a solid at high frequencies?** WR, PV

13. Further study is urged of a receiver unit with tubing coupling to the ear; also an electrostatic transducer lining a helmet. GC

14. The Harvintip should certainly be evaluated vs. the receiving end of the AIC/10 system. Also, the ANB-H-1A receiver in the Harvard Cushion should be evaluated vs. the AIC/10 system. PV, BBN

15. It has been suggested that a basic solution to the receiver problem would be an earphone small enough to be inserted in the ear canal, with a noise exclusion device placed over the entrance to the canal. GC

16. Refer to Baldwin AFAC-16 on acoustic performance of earphone, ear cushion, helmet combinations.

17. Considerable improvement should be possible in loudspeaker communication by means of:

a. increased power input,
b. closer placement,
c. higher efficiency,
d. greater clipping and optimized equalization. GC

18. In regard to use of an external loudspeaker: the operational feasibility must be evaluated in terms of power input and distance. If an earplug is to be used for attenuation, then the helmet itself might as well be made the attenuating device. One question is whether a useful attenuation spectrum can be achieved with a helmet. The primary question is whether an adequate signal-to-noise ratio can be established outside the head. GC

19. Regarding communication by loudspeaker: Appendix 7 and 8, pp VIII - using a loudspeaker with one watt input at a distance of 3 feet shows intelligibility in 98 db noise. Report AFAC-3, Figure 3 at one foot shows intelligibility in 115 db PN and 100 db JN.

20. Re loudspeakers: If for normal use a fairly close distance can be used, then for particularly critical conditions the helmet could be pressed against the loudspeaker forming a sealed cavity and hence increasing the effectiveness of the acoustical transfer. If necessary, a valve could be so arranged as to pressurize the cavity between the helmet and loudspeaker to the pressure altitude within the helmet, again improving the coupling. PV

21. Although performance of the AIC/10 system is admirable, it is probable that improvement in regard to other factors such as wearability may degrade the acoustical performance, in which case this would have to be restored. Hence with this anticipation improved performance must be an objective. For example, if a cushion may not touch the ear, this has vast implications in regard to transducer design. GC

1.3 Noise Reduction - Microphones

1. The gradient microphone of the AIC/10 system behaves similarly to the early carbon noise-cancelling units during the war in that the noise field is decreased until the mouth is opened; in other words, the articulation modulates the noise field. In fact, the modulated noise has some degree of intelligibility itself. It has therefore been suggested that the increased noise during articulation may be an assistance rather than an interference with speech. However, since the source of the modulated noise is external to the mouth and the speech has an internal modulator, it is inconceivable that these two sources would have any degree of coherence. Hence we suppose that the noise is destructive. PV

2. Many have stressed that there may be more to be gained by achieving improved noise exclusion than by improving the type of speech pickup. HF, CH, GC, NEL

3. We must recognize the importance of the helmet because it controls S/N at both ends of the system. Important aspects are attenuation, effect on the voice, acoustic modes inside the helmet influencing voice pickup, how to control these resonances. The following should be measured:

a. The attenuation of the present type helmet,

b. The voice response inside the helmet, and

c. What is the maximum attenuation that can be expected from a helmet? PV

4. We need data on the attenuation of the oxygen mask. Is there a difference in the apparent attenuation depending on whether a pressure or a gradient microphone is used? WS, WR

5. Improvements in a noise shield may be possible. It may be important that the microphone be flexibly suspended within the mask as near to the lips as possible Acoustical absorption may be added. PV

6. It has been suggested that if and when extreme noise or discomfort become insurmountable problems, then perhaps speech communication should be abandoned in terms of Morse Code in the form of a tactile signal on the arm or leg. An intermediate phase is to use the tactile

signal to warn when auditory communication is to be expected so that the necessary equipment can be applied, for example, a very uncomfortable noise shield. HF

1.4 Noise Reduction - Receivers

1. It is our general impression that helmet design is being pursued with too little experienced or competent acoustical guidance. GC

2. In connection with the limits of attenuation, the mechanism of bone conduction is important. From Fletcher's papers on Dynamics of the Middle Ear (JASA 24, 129, 52) and on the Dynamics of the Cochlea (JASA 23, 637, 51) it would be concluded that the mechanism conduction is an inertial one. Also the inertial sensitivity of a pair of ears should be relatively independent of the direction of the sound. HF

3. A useful reference is to Shaw and Yates (PAL) method for loudness balance method of measuring attenuation.

1.5 Evaluation Criteria - Physical

1. In regard to evaluation criteria: it is important in studying various methods of speech pickup that the evaluation technique should be diagnostic, explaining the reasons for the success or failure of a given system as well as simply giving an overall evaluation of relative performance. The diagnostic criteria should minimize uncertainties associated with personnel as in articulation testing. It seems likely that the associated physical tests may furnish diagnostic data. GC

2. Among the various physical tests, real voice frequency response should be measured and adjusted before any further testing is attempted, including articulation. In an electronic system overall response or real voice response can be so readily corrected by equalization that correctness in this respect is the least that can be expected. GC

3. The net overall evaluation in physical terms should be the speech-to-noise ratio, shown in our chart as dynamic speech/noise ratio by which we more properly mean speech/dynamic noise ratio. This measure as a function of frequency coupled through the articulation index would correlate with word articulation, but only if the speech is normal speech in terms of vowel-consonant ratio at the pick-up point. PV, WS, GC

4. Lacking correlation or in other words, for

diagnosis, one should measure consonant/vowel ratio, and speech-sound alteration (instead of distortion) properties. PV

5. It is questionable to what extent consonant-to-vowel ratio can be measured for consonant sounds. This may be possible for only a restricted group of consonant sounds, such as the fricatives. These at least have some duration and distinctive portion. Opinion was that we should be satisfied to get consonant vowel ratio only for fricative consonants. CH, GP, GM, JL, HL

6. In choice of word material for measurement of consonant-to-vowel ratio, much might be done by proper selection of words. For example, a short "i" vowel is both low in level and short in duration, as for example ;in the word "chick." CH

7. It must be realized that a "microphone" is always a combination of a transducer, coupler and noise excluder. The properties of interest are always for the combination and therefore evaluation has meaning only in terms of the combination. PV, WS

8. There is the large subject of talker/listener acceptability in terms of listenability, naturalness, pleasantness (lack of annoyance), speaker recognition, discomfort at high levels, comfort in terms of wearability. GC

9. In regard to evaluation criteria for speech reception, the measurement of masked threshold was added to the list after clarification. This has been a favorite technique both at PAL and RCA during the war. The confusing thing about this technique is that it is difficult to understand its function so far as any unique characteristic is concerned. Actually, it evaluates the combination of real ear response and cushion or helmet attenuation. In other words, if the motor mechanism is the same for two systems, the technique will evaluate attenuation, or visa versa, etc. One must also assume that the noise spectrum is the same. In terms of the calculation of word articulation, the masked threshold establishes a floor for the plotting of articulation index. RCA, DM

10. We have discussed coupler vs. real ear calibration of receivers. The real ear calibration shows a drooping characteristic as compared with coupler calibration. This is a characteristic of the ortho-telephonic concept, which includes the diffraction properties of the head. The difference is an extremely basic one in the overall response of systems. Pertinent references are:

JASA - 11, 278, 1940 - Dunn and White
JASA - 19, 90, 1947 - French and Steinberg

IRE - 35, 880, 1947 - Beranek
JASA - 22, 833, 1950 - Martin and Touger
JASA - 26, 679, 1954 - Burkhardt and Corliss.

DM, RCA, PV

11. In relation to the word envelope traces which PSV displayed to the Panel, the low frequency modulation in the envelope, even in the high frequency bands, was quite intriguing. Several possible explanations were suggested. One was that the high frequency components may be amplitude modulated by the low frequency components. Another was that the speech wave is not symmetrical and therefore that a full wave rectifier will produce a modulation at the fundamental rate. Another was that even when the low frequency components of a wave form are eliminated by filtering, the residual wave, consisting of a group of high frequency harmonics, will still have a wave length corresponding to the fundamental frequency. The latter proves to be the correct one. PV, WS, JL, GP

12. Some opinion favors ignoring a physical measure of S/N, but rather relying on articulation tests designed to stress consonants. HF

1.6 Evaluation Criteria - Articulation

1. It is generally felt that the whole subject of articulation testing could use a re-study. The major question is whether word lists can be optimized to provide greater efficiency in terms of information per unit time. Perhaps optimization must be for a specific purpose. GM, JL, GP, PV, WR, WS

2. Use of PB words for articulation testing is considerably limited in efficiency because too much of the score depends upon easily recognized sounds and the knowledge that the words are standard English words is a formidible clue to recognition, and hence the sensitivity of the test is diluted. Lists consisting of words which are often missed should make the test more sensitive and therefore increase the spread between various systems. GM, JL, GP, PV, WS, WR

3. Articulation testing in some form must be considered as the basic overall evaluation for exploratory systems in our program. GC

4. Presumably valid general principles regarding speech intelligibility are:

a. vowel sounds are very durable in terms of recognizability under various types of stress and

b. relative intelligibility of systems will therefore depend largely upon their ability to project consonant sounds.

If this is the case it would seem that the most efficient articulation testing would be accomplished in terms of consonant recognition. GM, JL, GP, PV

5. Study of the confusion tables of G. Miller's paper (JASA 27, 338, 1955) will find the most difficult consonants. **GM, WS**

6. We have had considerable discussion of the question "how do we recognize consonants?" Certainly not entirely on the basis of consonant energy alone, as if one could assign a definite time interval to the consonant alone. In many cases, such as the stop consonants or plosives, no specific time interval can be attributed to the consonant. In all cases there is a mutual influence between the consonant and the following vowel sounds. The building blocks of speech are apparently not isolated or unique entities (phonemes). Coupling is an all-important factor. (See Harris, JASA 25, 962, 1953) CH, GP, HL, HF, WS

7. The vowel energy is not what we want to measure. We want the consonants and their influence on the vowel. In many cases this influence may be the sole cue to the consonant intelligibility or recognition. GP, CH

8. G. P. demonstrated samples of speech with either vowels or consonants removed. Neither is intelligible. He then demonstrates a sentence which is spliced together from discreet consonant and vowel sounds. This is also completely unintelligible. It may be questionable whether this synthesis was accomplished with the precision of phoneme-duration and relative levels which may be required. In any case it illustrates thoroughly the fact that intelligibility is greatly dependent upon the transitions from one speech sound to another. It certainly cannot be assumed that speech sounds are unique, self-sufficient entities which can be placed in succession to create intelligibility. GP, CH

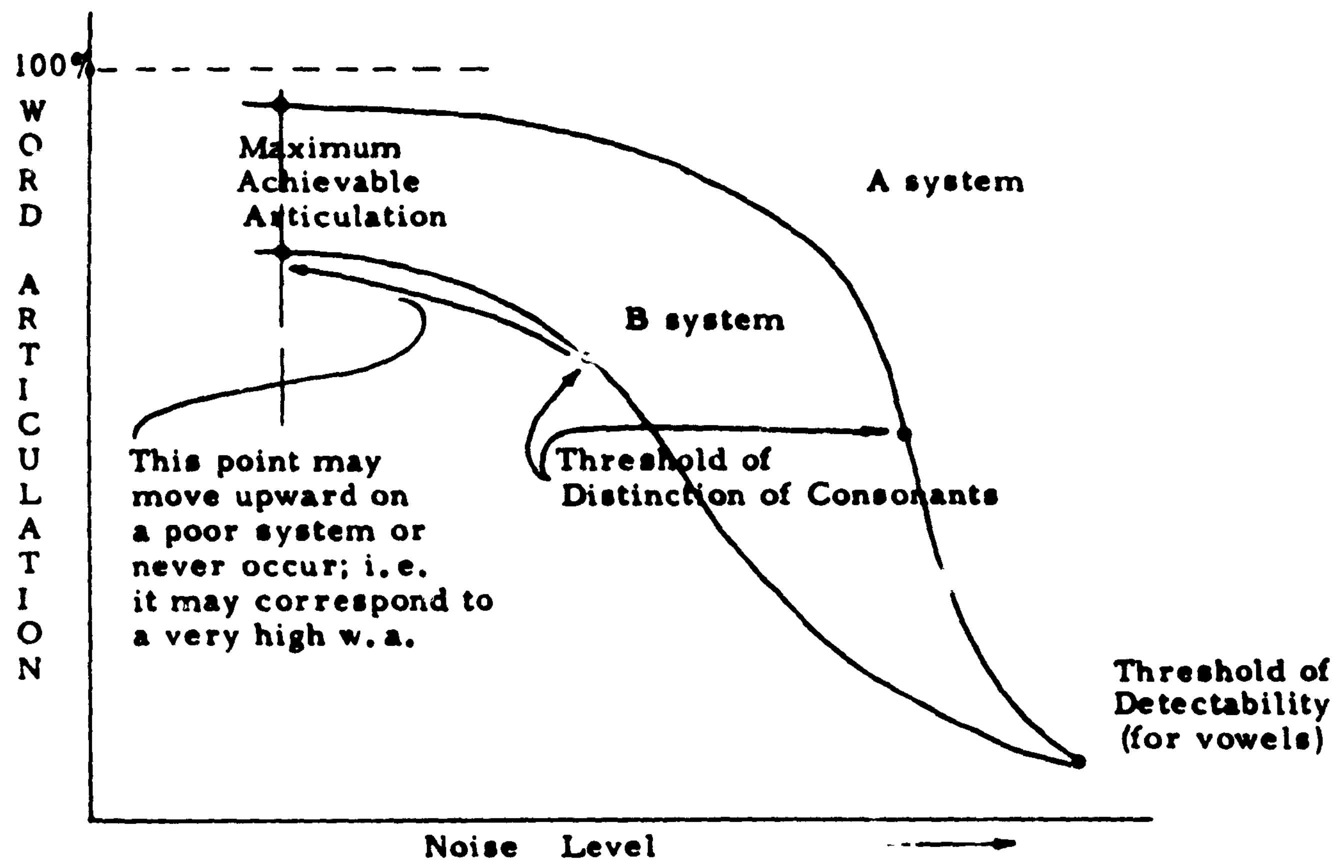
9. An important basic study would be to find where the significant part of the consonant or word is by:

(1) cutting back on the word until the remainder is completely masked in the noise background, hence the remainder cannot be significant or,

(2) cut the initial part of the word until the difference is distinguishable. In this way one could presumably find out the relative importance of the static vs. the transitional portion of a consonant sound. It is at a signal-to-noise ratio of about + 10 db, the static portion of consonants can be contributing greatly to intelligibility when they are probably at a signal-to-noise ratio to the order of -10 db, and being of rather distributed spectrum, are probably completely masked. Therefore, one is inclined to suspect that the transition regions must be the principle cue to consonant sounds. PV, GP

10. We discussed at length the possibility of using a continuous series of spoken syllables using the same vowel with several starting consonants and determining the

signal-to-noise ratio at which the listener can distinguish the difference between the syllables, in other words, distinguish the consonants. This test may be particularly effective by using a low level vowel of short duration. This may be better restated as follows: What we are really looking for is a short single point test which will measure the noise level in which a system will achieve some fairly high degree of intelligibility. We wish to find a test which will rank-order systems properly in regard to excellence of speech articulation. It may be that a given system can never achieve high articulation by virtue of limited frequency range or poor consonant pickup. If this were so it would be evidenced by a test based upon recognition of consonant sounds where it might never be evidenced by recognition of vowel sounds. The proper diagram for illustrating this test is shown below.



The test would work as follows: For a fixed voice effort level, noise to quiet test, and a fixed favorable listening level for the listeners, the noise level surrounding the speaker would be gradually decreased until the listener achieved the ability to recognize and distinguish the words by virtue of hearing the consonant sounds. The noise level at which this occurs is presumably at a fairly high value

of word articulation. It may not necessarily be the same percentage word articulation for all systems. It is presumed that systems rank ordered in this manner would have the same relative rating as if they had been rank ordered in accordance with a complete and more conventional study. PV, GP

11. Now the question is, how can the effectiveness of this proposed system be proved? Presumably one could perform the test indicated in the above chart using one excellent system and another system which is stressed in some particular way, such as limited frequency response or by poor consonant pickup. The complete curves should be traced out for each system and the threshold of detection for words and then for the special consonant word list performance. If the points turn out to be oriented as supposed by the chart, the method would have been proved, at least for these two systems. PV, GP

12. The task of evaluating ten microphone systems by standard word lists has been evaluated: use five talkers, six listeners (crew of seven) six points on S/N curve, each point 10 word lists. Hence 300 word lists per system = 56 hours per system = 392 man-hours per system. Hence for 10 systems - 4000 man hours = 2 man years. Hence the need for an abbreviated technique is evident. DM

13. There is considerable approval of use of CVC word lists, choosing one vowel and using all the consonants. Vulgarities must be eliminated. GP, CH, JL, GM, HL, PV

14. Much might be done by proper selection of words to stress consonant-to-vowel ratio by duration as well as level. For example, a short "i" vowel is both low in level and short in duration, as for example in the word "chick." CH

15. The mutual influence of consonant and vowel is such that by proper choice of vowel sound the consonant sound may be enhanced. CH

16. Assuming that there are 25 possible initial consonants and 21 possible final consonants, there would be 525 possible CVC words using only one vowel. If all 15 vowels and diphthongs were used there would be 7875. Many of these cannot be spoken. With such a list on cards thoroughly shuffled, one would not need formal word lists. Should the carrier sentence be used? Should the words be written? How to avoid wasted time for grading? If we concentrate on the fricative consonants and choose only one vowel, there would be 8 x 2 = 16 CVC words. This same word list can be used for the physical measurements. Question is whether the consonants would be properly represented when used with a single vowel. Regarding choice of vowel, see

Fletcher pps. 87, 63 and 2. A vowel can be chosen from the central region. Harris concludes that three vowels are enough. Our tentative conclusion from all this was to use the simplest possible test for rank ordering until the differences are too small to be distinguished by the accuracy of such a simple test. For this purpose we propose a short word list such as the 16 CVC (fricative) words spoken in random sequence without carrier sentence, noting only the noise level in the surrounding space for a threshold of distinguishability. For more accurate testing, we would use the longer CVC word list with the carrier sentence, hoping to arrange for a self-grading scheme. GC

17. Another test which was proposed was to read newspaper material continuously and to adjust noise level for the threshold of intelligibility. Will rank ordering be different for different talkers? JL

18. The suggestion has been made that whispered speech might be of value in articulation testing. There is considerable interest, but it is not favored because it is an unusual manner of speaking. RB, GP, GM, JL

19. It has been asserted that most efficient articulation testing is with talker-listener pairs, especially when the microphone end is suspect. Reason is that talker efficiency is such an important factor in the result. JL

20. There is a testing method which automatically finds 50% (or any other) articulation point: each word rightly heard decreases gain 1 db, each word wrongly heard increases gain 1 db. JL

21. Circular articulation testing has been suggested: a word is started and repeated by each listener. Result is to sharpen articulation vs S/N curves, presenting $(W.A)^n$. Question is how this technique can be used in a noise-to-quiet test. JL

22. For rapid testing the process may be simplified by uncovering successive words in a list, the listener simply marking "right" or "wrong" as heard, hence self-grading. BBN

23. In the interests of repeating words or sentences consistently, quickly, or with good diction, it may be well to use a cuing record which is played into the speaker's ear just before he repeats the sound. CH

24. A technique for monitoring speaking level for each talker is as follows: using a VU meter, find the output level for maximum effort for a given microphone system. This will vary from system to system depending on the encumbrance of the voice by enclosures. Having found the level for maximum effort for a given system, it is reduced 6 db and this voice level is held. WR

25. The IC-10 microphone must be included along with the exploratory systems, because performance must be measured in relation to this system. GC

1.7. OTHER EVALUATIONS

1. There is the large subject of talker/listener acceptability in terms of listenability, naturalness, pleasantness (lack of annoyance), speaker recognition, discomfort at high levels, comfort in terms of wearability. GC

2. Evaluation of transducers must also be in terms of unusual feed requirements or shock voltages. RCA

3. There may have been a wrong trend in AIC/10 development, in having attempted to achieve a flat overall orthotelephonic response with a 6 db/octave response from the microphone and the reverse in the headset. This original concept was in terms of use of a bone conduction receiver. The change to an air conduction receiver changed this concept. The real ear response of the receiver (Figure 5 of Snow summary) shows a rising real ear response for the receiver as well, hence accentuating the high end of the microphone - giving very disagreeable quality. C & N Lab. considered this to be essential for high intelligibility. DM

1.8. ELECTRONIC TREATMENT

1. On the morphological chart another excellent reason for speech clipping should be added, namely protection of the ear against pain or discomfort of overload from too loud talking. RCA

2. Discussions have stressed that clipping is of value only when an adequate signal-to-noise ratio has been established at the microphone. (WS, PV)

3. On the basis of work with deaf children, frequency selective compression has been suggested. This is elaborate, requiring separation of spectrum into frequency bands. RB

4. The use of the Scott "dynaural" suppressor for suppressing noise in a clipping system has been suggested. This system may also be used to change equalization with level so as to leave the low end in the speech spectrum when the power capacity of the receiver units is adequate. WS

5. In regard to equalization: the question is whether it should be adapted to the noise conditions at the microphone end or the receiving end of the system. So far as the sending end is concerned, it will do no good for intelligibility unless the S/N is adequate. If the S/N is adequate, then it may be helpful in extreme cases to equalize the speech spectrum so that at the listening end the speech lies within the residual hearing area. There is in general an upper limit to the usable hearing area, generally definable by pain, tickling feeling, or extreme annoyance. This area for the moment may be assumed to be flat as a function of frequency and a an overall level of 130 db spl. The noise spectrum at the ear will define a lower boundary of usable aural area. As the noise level increases the

residual useful area may become very small. Hence,there will be an optimum shape of speech spectrum, presumably requiring maximum speech clipping, to make optimum use of the residual hearing area as a function of frequency. In other words, a typical speech spectrum might protrude into the residual hearing area only over a narrow frequency range, whereas with proper equalization it could be fitted into the residual area across the entire frequency band. PV

6. Referring to the possibility of frequency division and clipping, which is suggested in the interest of reducing distortion, it has been suggested that this can be accomplished at a microphone by using a two-way microphone, one for high frequencies, the other for low frequencies, the low frequency unit being equipped with diaphragm stops to accomplish the clipping mechanically. RB, WS

7. There has been considerable discussion of compression, or variable gain, in contrast to clipping. There is still some difficulty in concept involved. We must distinguish between compression, delayed compression, compression with time constant, and clipping. In regard to clipping vs instantaneous variable gain: if O is output, I input, G is gain, then for fixed gain $O = G I$. $G = \log I/I$. In general, the variable gain amplifier should have less violent distortion than a clipping system. From this point of view a clipping system is simply one form of instantaneous variable gain amplifier in which G is constant and zero above some value of O. PV, WS, RB

8. It is believed that the virtues of compression and clipping have been well used in the AIC/10 system. GC

9. The concept of a uniform speech spectrum has been favored. Instead of differentiation which decreases the low frequency level while increasing the high frequency level, the difference consists in simply raising high frequencies level so as to produce a flat real voice spectrum. Any virtue would presume an adequate S/N at the microphone. DM

10. There is some feeling that in order of priority no effort should be devoted to the electronic system until the microphone and receiver end have been clarified. This is because the value in any electronic treatment is contingent upon having established an adequate S/N at the microphone. GC

1.9. ALTITUDE COMPENSATION

1. The virtue of automatic altitude compensation built into the receiver unit of the AIC/10 has been questioned. There are so many other things which change with altitude, even vocal output, that it would seem preferable to find the overall function and provide automatic pressure actuated overall compensation. To build compensation into the receiver alone is a compromise, since efficiency suffers and also the acoustic source impedance is raised so that output level is more sensitive to the volume of the cushion. It has also been questioned whether the required compensation may be different for vowels and consonants. GC

1.10. LIMITS OF THE PROGRAM

1. The objectives of the program are sufficiently broad in statement that some limitations must be chosen. First we shall exclude coding devices, i.e. we assume that the program is to deal with direct voice communication. GC

2. On this basis we exclude teletype as a possibility unless an extremity of noise field demands. GC

3. Interesting possibilities have been suggested. There is evidence (Prof. Roman Jacobson - Language Dept., Harvard University 1949, "On the Identification of Phoneme Entities"), that a code could be devised from the position of lips, tongue, jaw movement, etc; George Miller, JASA 27, 338, Mar. 1955 "An Analysis of Perceptual Confusions Among Some English Consonants" suggests that a code could be devised from a classification of consonants in terms of voicing, nasality, affrication, duration, place. BBN, GM

4. Another interesting suggestion is the use of an intermediate dummy voice located in a noise-isolated box. The idea is to take mechanical position code from the real voice and feed it through a position servo to the dummy voice which does the actual speaking, which should then be free of noise interference. JL

5. An objective view of the present situation will undoubtedly conclude that the greatest single shortcoming of the AIC/10 system is in regard to wearability. While this subject should be vigorously pursued, it is disclaimed as an assignment for our particular program, even though our work is intimately dependent upon solutions to this problem. GC

APPENDIX 2.0

SUMMARY OF CHARACTERISTICS OF AN/AIC-10 INTERCOMMUNICATION SYSTEM

BY

W. B. SNOW

APPENDIX 2

SUMMARY OF CHARACTERISTICS OF AN AN/AIC-10 INTER-COMMUNICATION SYSTEM

Figures

APPENDIX 2.0

SUMMARY OF CHARACTERISTICS OF AN/AIC-10 INTERCOMMUNICATION SYSTEM BY W. B. SNOW

2.1 INTRODUCTION.

The panel has been furnished with three documents (Refs. 1, 2, 3) which give a good summary of the general physical form of the present system, together with comments on initial field experience. The additional documents represented by Refs. 4, 5, 6 and 7 add up to a 5 1/2-inch thickness of reports. These contain a wealth of interesting detail regarding the study, development and test of the system, comprising more than 65 man years of effort. They inspire admiration for the breadth of the studies and the pains-taking work of devising and engineering components that would meet the rigorous requirements for performance, flexibility and reliability which had been set.

The purpose of this memorandum is to give a brief resume of the main operating principles and measured performance of the AN/AIC-10 components as used with oxygen mask and hard helmet. This system is believed to be the closest present approach to a system capable of operating under the severe conditions the panel is to consider.

2.2 OVERALL SYSTEM OPERATION

The full system contains very flexible switching facilities which are necessary but do not affect acoustic performance directly. The fundamental system is as follows:

> A moving coil microphone of the noise-cancelling type picks up the speech very close to the talker's lips. The microphone output is amplified in an amplifier equipped with automatic gain control and supplied to the interphone line circuit at relatively constant level. During talking periods side tone is furnished in the talker's earphones. The microphone is connected only when the talk-switch is operated.
>
> At each receiving station the listener's earphones are connected to the interphone line circuit through an amplifier

equipped with a gain control and peak clipping circuitry. The listener can adjust the listening level as he desires, and the amount of peak clipping depends on the gain setting he selects.

In the discussions of details which follow, the characteristics are representative and approximate, but not always from one source, or upon identical instruments, since they were culled from various reports. However, they certainly show the main features of the present system.

2.3 MICROPHONE

The microphone of interest here is the M-32 (pg. 17, Ref. 3). It is a moving coil unit with a front and a back opening for sound, each covered by a moisture barrier. Figure A2-1 shows the axial noise discrimination of this microphone (Fig. 1-33, Ref. 4). Figure A2-2 is the real-voice frequency characteristic of this microphone mounted in an A-13A oxygen mask, at sea level (Fig. 2, Ref. 5). This was obtained by determining the integrated output in many frequency bands for a crew of talkers, compared to the corrected output of a Western Electric 640AA microphone on the voice axis in an anechoic chamber for the same crew. Figure A2-3 shows the change in microphone characteristics between sea level and 40,000 ft. altitude. (Fig. 1-16, Ref. 4).

2.4 EARPHONES

The earphones (H79/AIC = H75A in helmet, page 19, Ref. 3) are moving coil instruments in which great effort was expended to achieve a thin mechanical structure. The diaphragm motion is so controlled by an acoustic chamber that the sensitivity is substantially constant with altitude; the high frequency cutoff decreases with altitude but remains above 4000 cycles at 40,000 ft. Figure A2-4 shows the frequency characteristic of the earphone with closed coupler, and with a coupler load simulating that of the earcushions, vs. altitude. (Fig. A-1 and A-2, Appendix II, Ref. 4). While this shows 98 db response, recent production is running around 102-103 db because of improvements recently added. Figure A2-5 gives the real-ear calibration of the earphone, defined as "- - - the sound pressure level of a sound wave approaching a listener (ears uncovered) facing a sound source in a free progressive sound field, which yields the same loudness sensation as the tone which would be heard by the listener wearing the headset with the specified voltage applied to the headset terminals." (Fig. 1, Ref. 5). Non-linear distortion is shown by Figure A2-6 (Fig. A-21, Appendix 2, Ref. 4).

The earphone unit is mounted in a cushion composed of a molded contoured nylon backing shell, dipped neoprene front skin filled with specially shaped filler ring, and thin nylon cloth

cover (page 19, Ref. 3). It is stated that "the bulk of the packing surrounds the pinna, placing the largest part of the ear cushion pressure on the head." It is an "earcap," however, not a doughnut design surrounding the ear and bearing only on the head. It is stated to be satisfactory for from two to four hours continuous wear, but for longer periods cords are provided to withdraw the unit from the ears when desired. Normal pressure is 20 - 30 ounces, but noise exclusion is unaffected down to 15 ounces. The cavity volume is about one-fifth that of the old standard M301 doughnut cushion. Figure A2-7 gives the noise exclusion for the earphones in a headband (Fig. [illegible]-3, appendix 2, Ref. 4). This is essentially the same as in the helmet up to 2000 cycles; above this frequency the helmet combination yields an additional exclusion reaching about 10 db at 6000 cycles. (Ref. 5).

2.5 [illegible]

The talking and listening amplifiers are actually the same unit, with function changes made by the talk relay. The frequency response is down 2 db at 300 and 6000 cycles.

5.1 In talking mode, the maximum power gain is about 70 db for inputs below -55 dbm. Above this input the automatic gain control becomes effective and the output rises only 5 db as the input increases to -25 dbm. Nominal normal output to the interphone line is +17 dbm at about -45 dbm input. The harmonic distortion at -25 dbm input (approximately +21 dbm output) is less than 7[illegible] and falls below 2[illegible] at -35 dbm input. Automatic Gain Control attack time is less than 0.1 second; release time is 17 seconds.

5.2 In listening mode the gain is slightly lower at maximum and controlled by a gain control with 35 db range. No automatic gain control is used, but the circuits are arranged to clip peaks symmetrically without blocking. The overall connections are such that with full gain peaks are clipped about 15 db. It was found that listeners preferred a setting giving 5 db or less of clipping in 120 db jet noise. The harmonic distortion in listening mode is 3[illegible] at +18 dbm output, rising to 27[illegible] at +27 dbm output. The listening amplifiers are connected with series padding resistors so that they can be bridged to several outputs at once without producing crosstalk. This protection amounts to 94 db in the final design.

2.6 [illegible]

The [illegible]aldwin Company made various laboratory tests in simulated aircraft noise with spectra as shown in Figure A2-8 (Fig. 3, Ref. 6).

6.1 The combined noise and speech output, and noise-alone output of the [illegible]-32 microphone in A-13A Oxygen mask is given in Figure A2-9 (Fig. 17, Ref. 5), for 120 db overall level jet noise, while the

difference between the curves, or the signal-to-noise ratio, appears as Figure A2-10 (Fig. 18, Ref. 5).

6.2 Figure A2-11 shows the masked threshold level for the H75/AIC headset in a helmet in the 120 db jet noise (Fig. 12, Ref. 5). Zero on this scale represents about 70 milliwatts into each earphone, or +18.5 dbm single frequency. The approximate threshold without noise is shown for comparison.

6.3 Figure A2-12 (Fig. 9-26, Ref. 4) illustrates the composite effect of the final noise contributions for components from microphone and earphone, in relation to speech levels from the microphone. These data were part of an early study, but they illustrate the general relations. They show that at low frequencies the principal noise interference comes in at the listening end, while at high frequencies it gets into the system via the microphone. The calculated word articulation from this plot was 64%. As appears below, the measured value of the nearly equivalent system, but with somewhat more high frequency noise exclusion at the earphone and because of the helmet, was 68%.

2.7 WORD ARTICULATION PERFORMANCE

The Baldwin Company made word articulation tests on a production AN/AIC-10 setup with M-32/AIC microphones in A-13A Oxygen Masks, and H-75/AIC Headsets in P-1A Protective Helmets (Ref. 5) in simulated airplane noise. The noise spectrum was that of the "130 db system" of Figure 8, adjusted to overall levels from 120 to 128 db. For these tests, as the result of some preliminary ones, the receiving volume controls were set at full gain, which was acceptable to the observers. Crews were experienced. Two talkers read two [illegible] Word Lists each for each noise level, to six observers. The word articulation scores are shown in Figure A2-13.

REFERENCES FOR APPENDIX 2

1. Intercommunication Set A[illegible]/AIC-10, Memorandum by [illegible] -3 [illegible]L/J1,12 [illegible]ember 1952 (distributed to panel).

2. [illegible]emorandum on Intercommunication Set A[illegible]/AIC-10, [illegible] - L-[illegible], 4 December 1953 (distributed to panel).

3. Intercommunication System AN/AIC-10 Operating Instructions and Maintenance Reference RCA (distributed to panel).

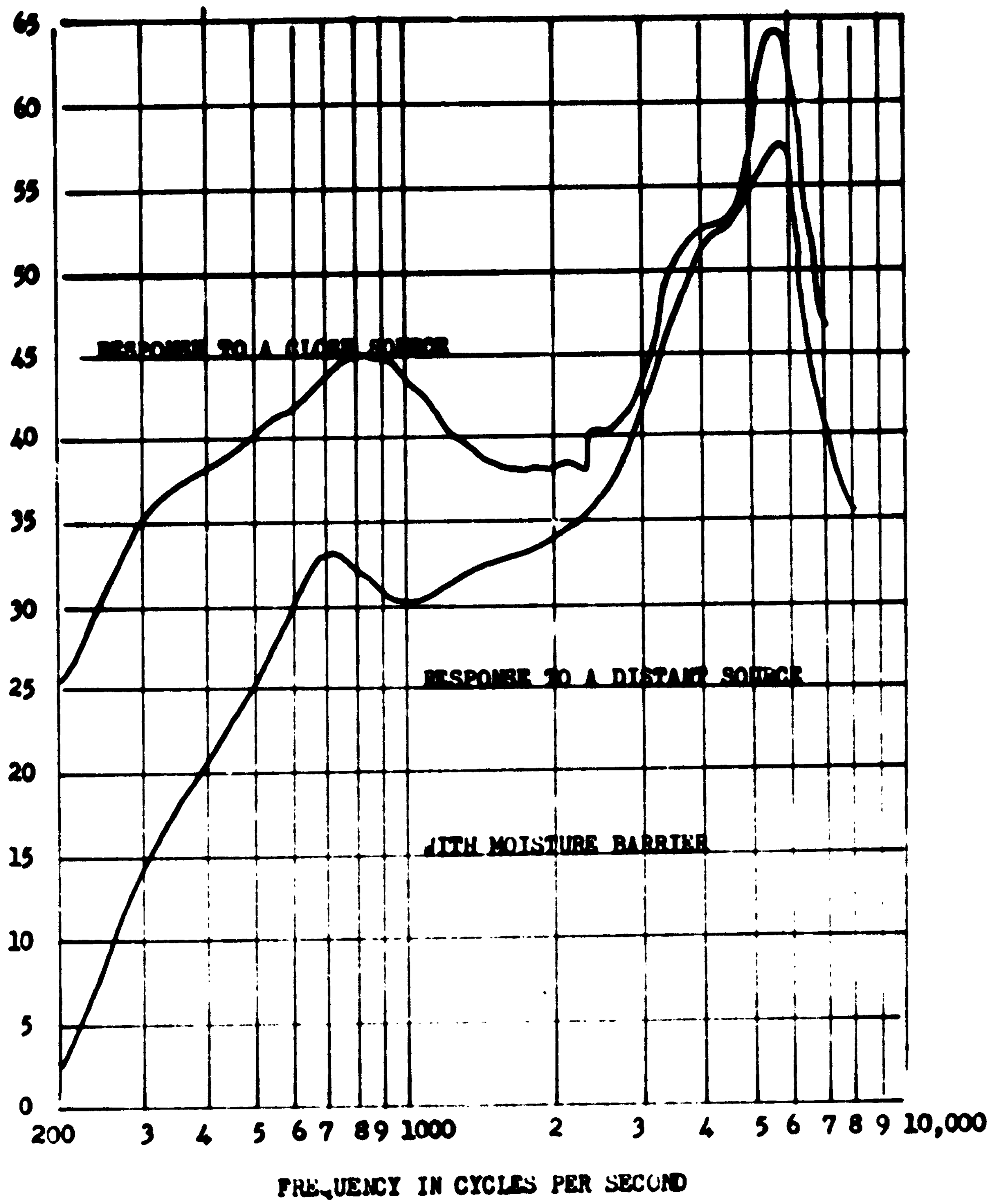
4. [illegible]/AIC-10 ([illegible]A) High Intelligibility Interphone Equipment, Final Report, Contract W-33-038-ac-18181, [illegible], [illegible] - Special [illegible] Engineering, Radio Corporation of America

   Part I - Summary
   Part II - Recommendations
   Part III Appendix I. Microphones
   Appendix II. Earphones, [illegible], Connectors and Cables
   Appendix III. Amplifiers and Control Units
   Appendix IV. Dynamotors
   Appendix V. Standby Loudspeakers
   Appendix VI. External [illegible] [illegible]
   Appendix VII. Installation [illegible] and Problems
   Appendix VIII. Performance of [illegible]lectrical Units
   Appendix IX. Systems, [illegible] and Performance
   Appendix X. Bibliography

5. Report A[illegible]C-10 [illegible] [illegible] [illegible]/AIC-10 Intercommunication Set, Final Engineering Report on [illegible] 10, Contract [illegible]33(038)-2[illegible]12, [illegible] 195[illegible], Engineering and Research Department, The [illegible] Company, Cincinnati 2, Ohio.

6. Report [illegible]C-11 - A[illegible]AIC-10 Articulation [illegible] in [illegible] High Ambient Noise - Final Engineering Report on [illegible] 11, Contract [illegible]33(038)-2[illegible]13, [illegible] 1953, Engineering and Research Department, The [illegible] Company, Cincinnati 2, Ohio.

7. A Bibliography of Speech Communication in Noise, [illegible] Report [illegible] Issued under Signal Corps Contract [illegible], [illegible] of [illegible] in [illegible] Ambient Noise [illegible].



AXIAL NOISE DISCRIMINATION OF AN M32 MICROPHONE

Figure A2-1

REAL VOICE RESPONSE-FREQUENCY CHARACTERISTICS

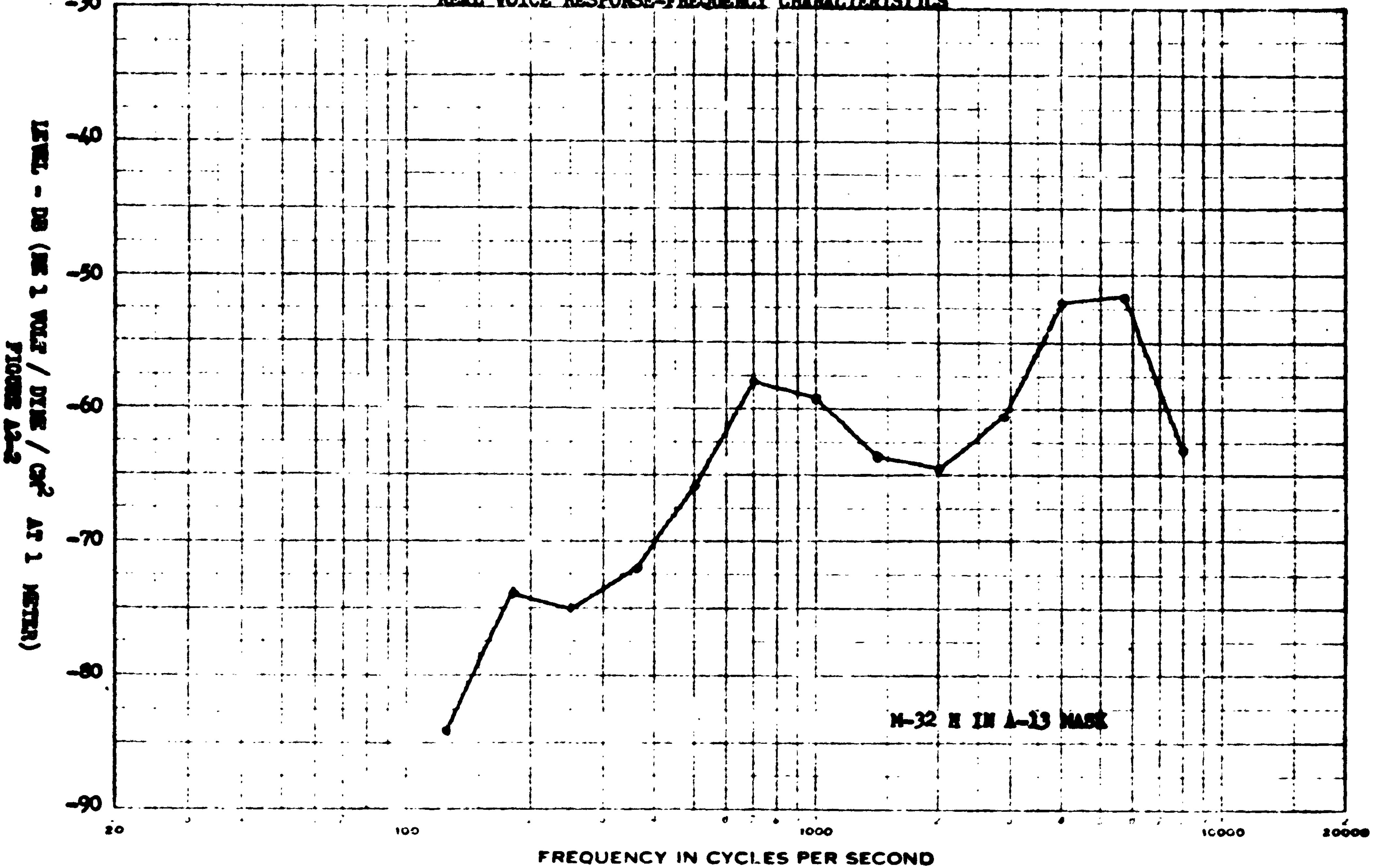
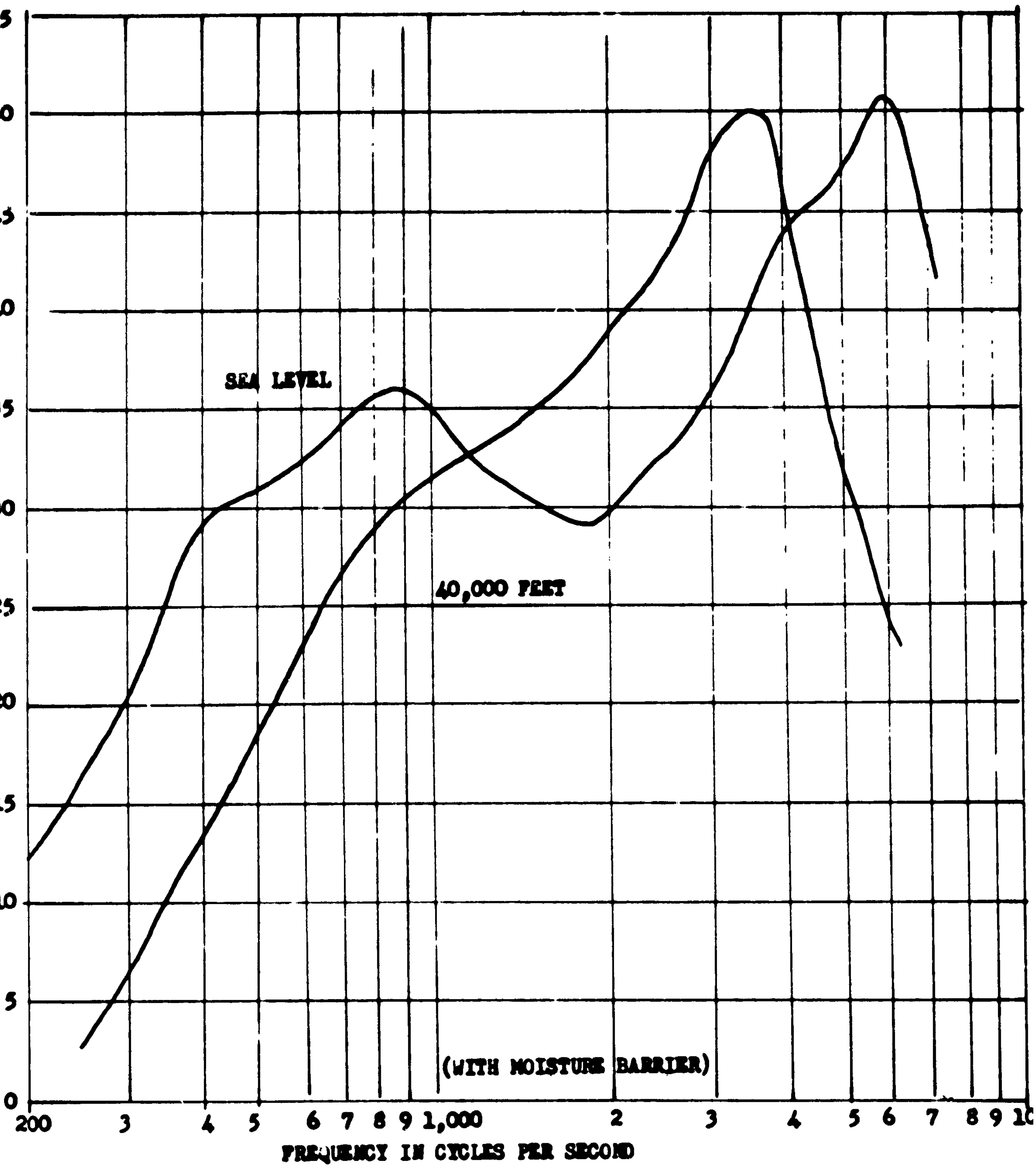


FIGURE A2-2

FIGURE 2

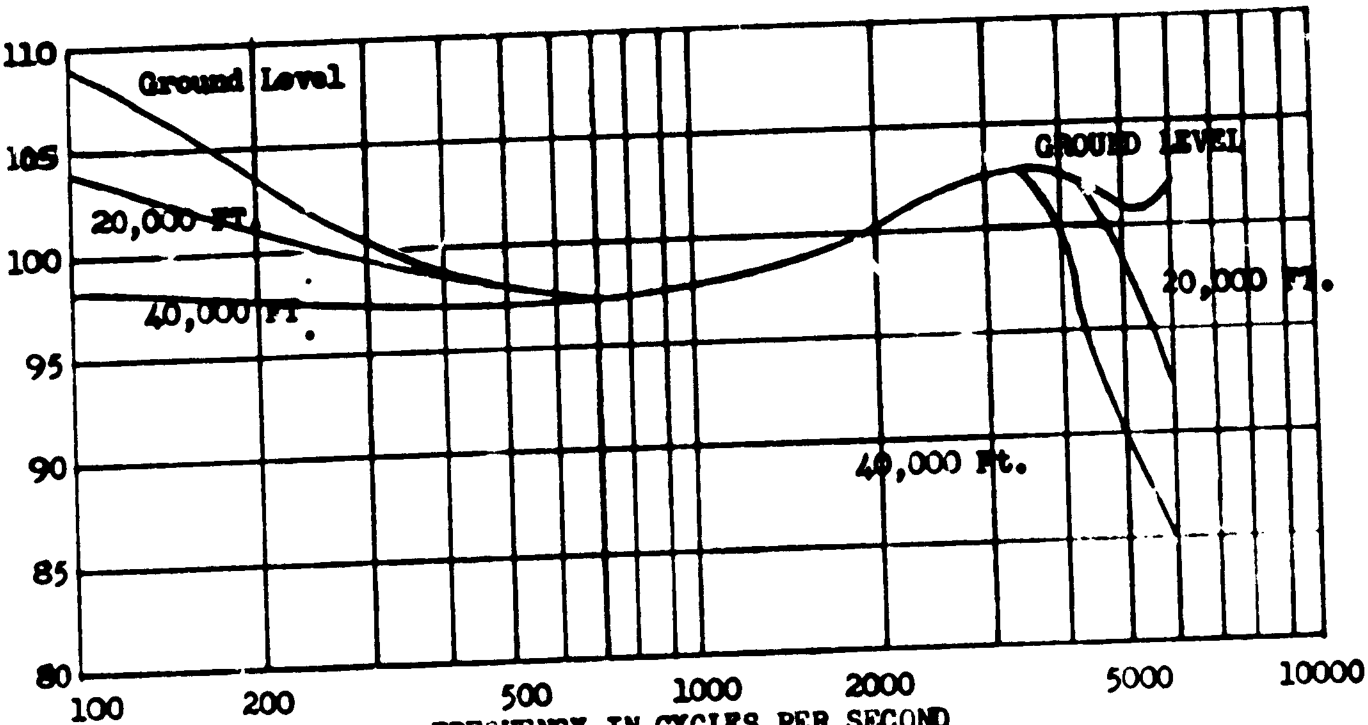


FREQUENCY RESPONSE OF A TYPICAL M32 MICROPHONE AT
SEA LEVEL AND 40,000 FEET

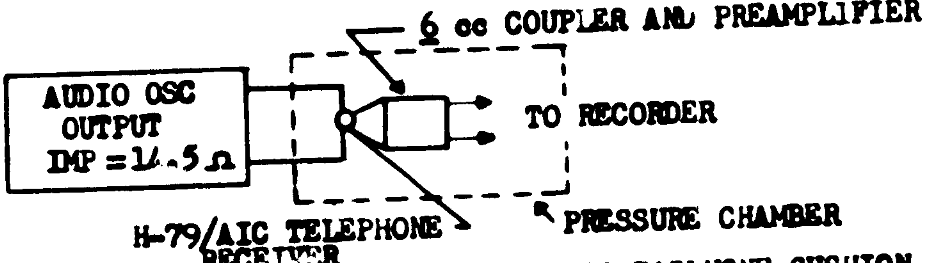
Figure A2-3

ALTITUDE RESPONSE FOR H-79/AIC TELEPHONE RECEIVER
(DEVELOPMENT MODEL)

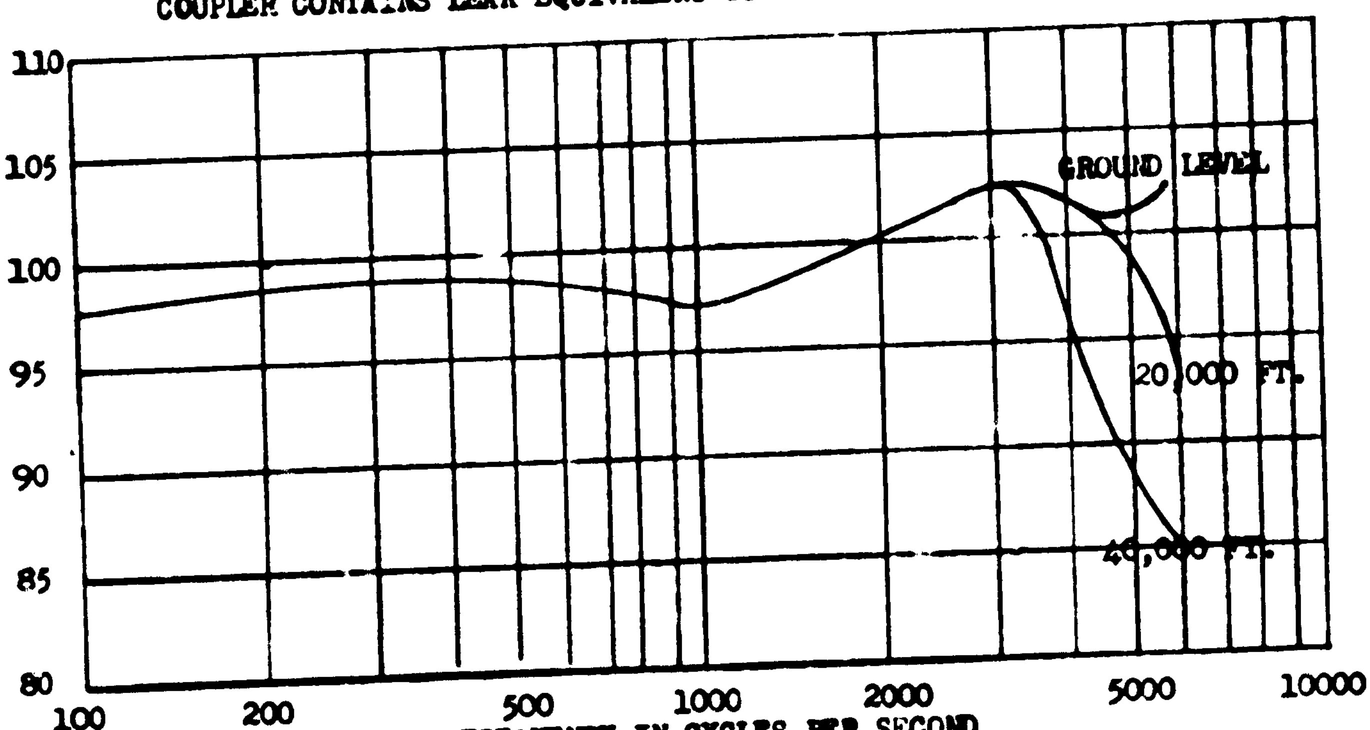
CLOSED COUPLER



FREQUENCY IN CYCLES PER SECOND



COUPLER CONTAINS LEAK EQUIVALENT TO EARPHONE CUSHION EFFECT

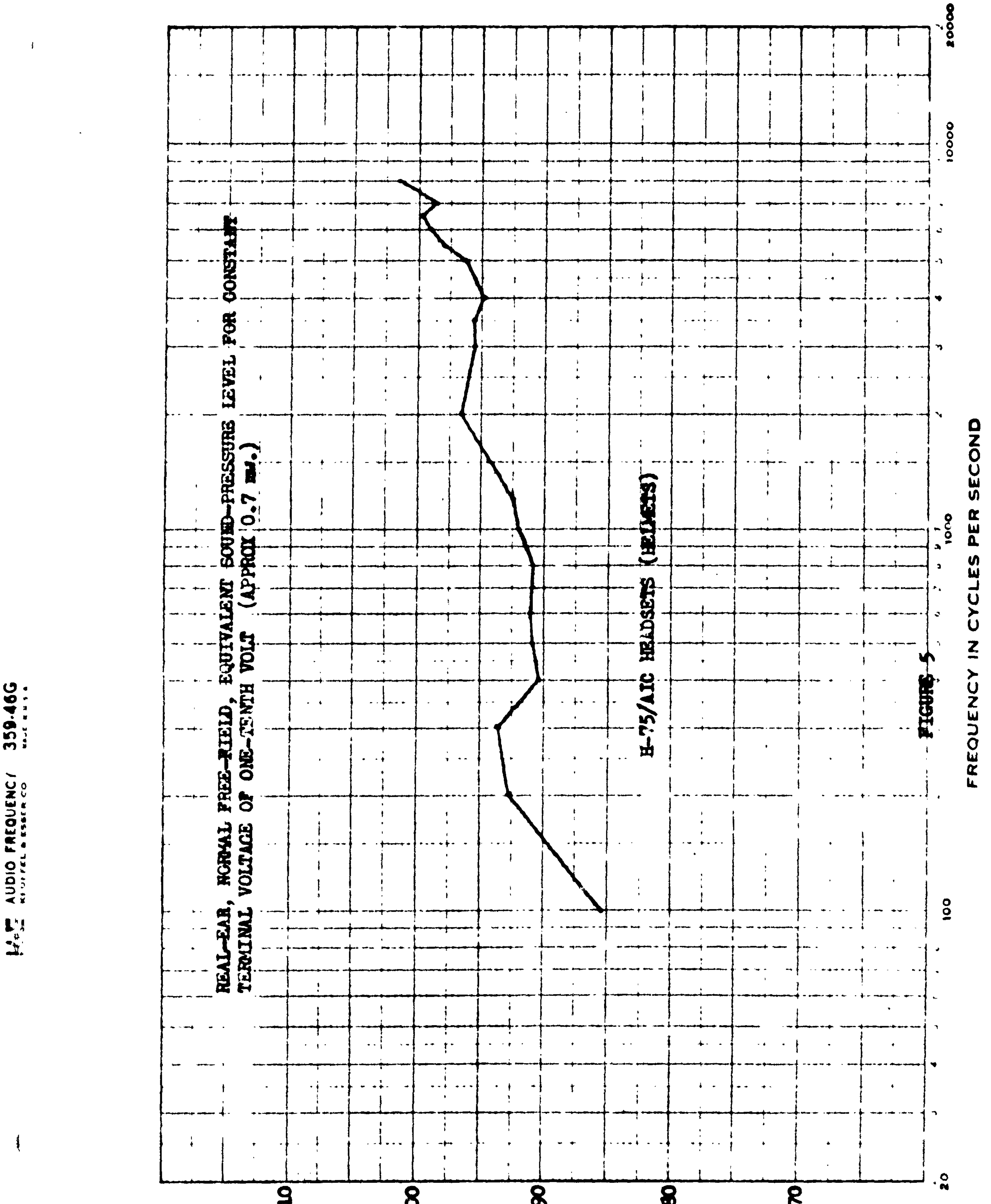


FREQUENCY IN CYCLES PER SECOND

NOTE (1) 1MW CONSTANT POWER AVAILABLE FOR 14.5 OHM EARPHONE

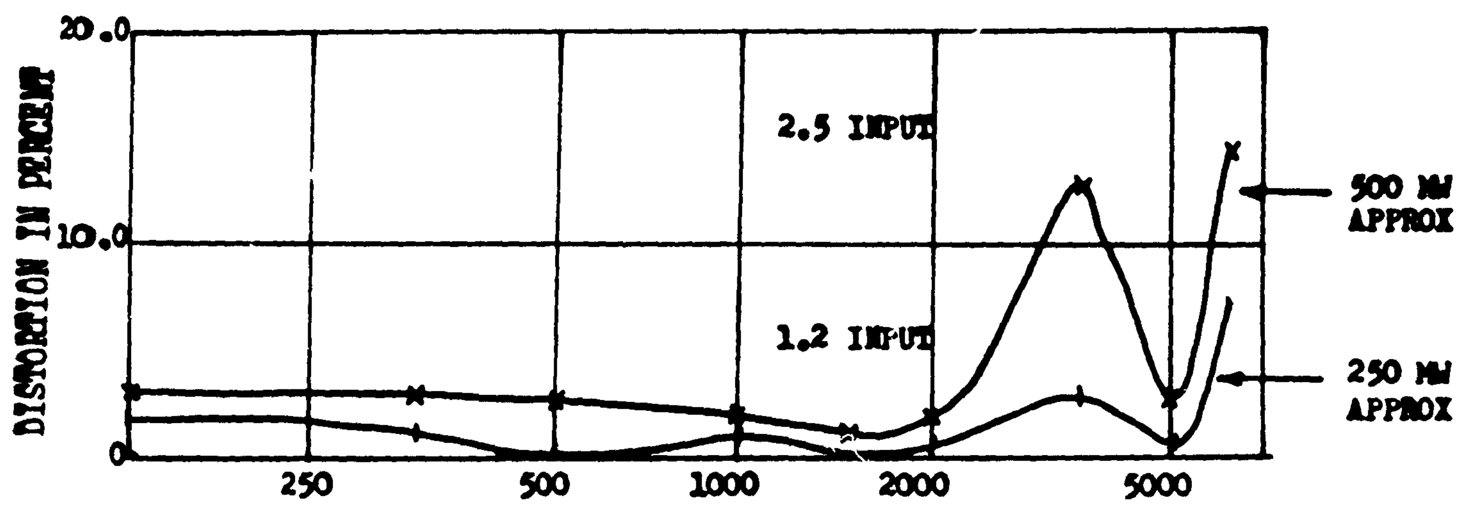
NOTE (3) 20,000 FT. PRESSURE = 349.1 MM HG
40,000 FT. PRESSURE = 140.7 MM HG

FIGURE A2-1



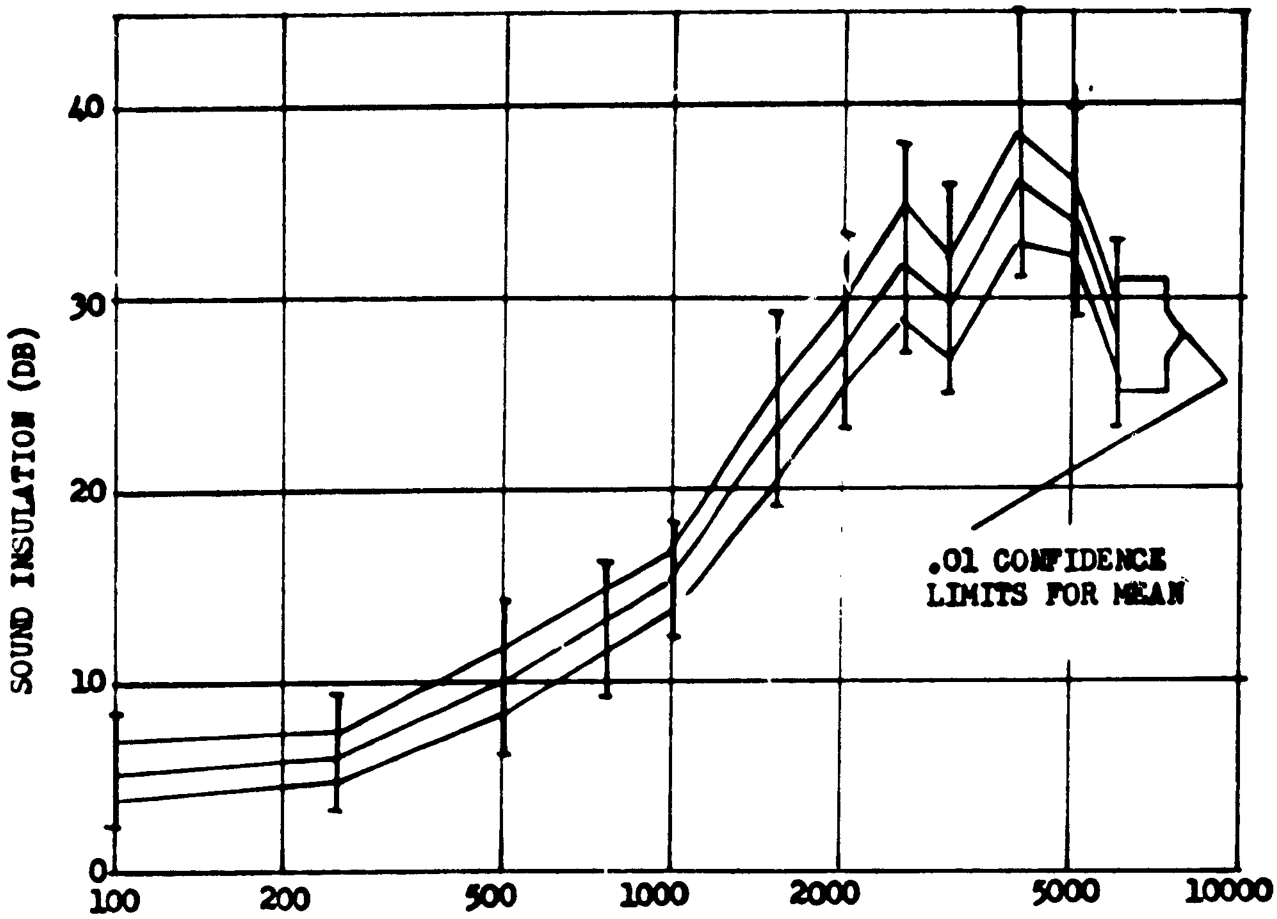
EQUIVALENT SOUND PRESSURE DB (RE .0002MICROBAR)
FIGURE A2-5

5



TELEPHONE RECEIVER H-79/AIC
PERCENT HARMONIC DISTORTION
(INITIAL PRODUCTION DESIGN)

Figure A2-6



FREQUENCY IN CYCLES PER SECOND

SOUND INSULATION HEADBAND HEADSET, H-70/AIC LOUDNESS COMPARISON - 100 DB FREE FIELD FOR HEADSET ON 7 SUBJECTS, 2 TESTS PER SUBJECT

FIGURE A2-7

SIMULATED AIRCRAFT JET NOISE SPECTRA- BALDWIN NOISE CHAMBER

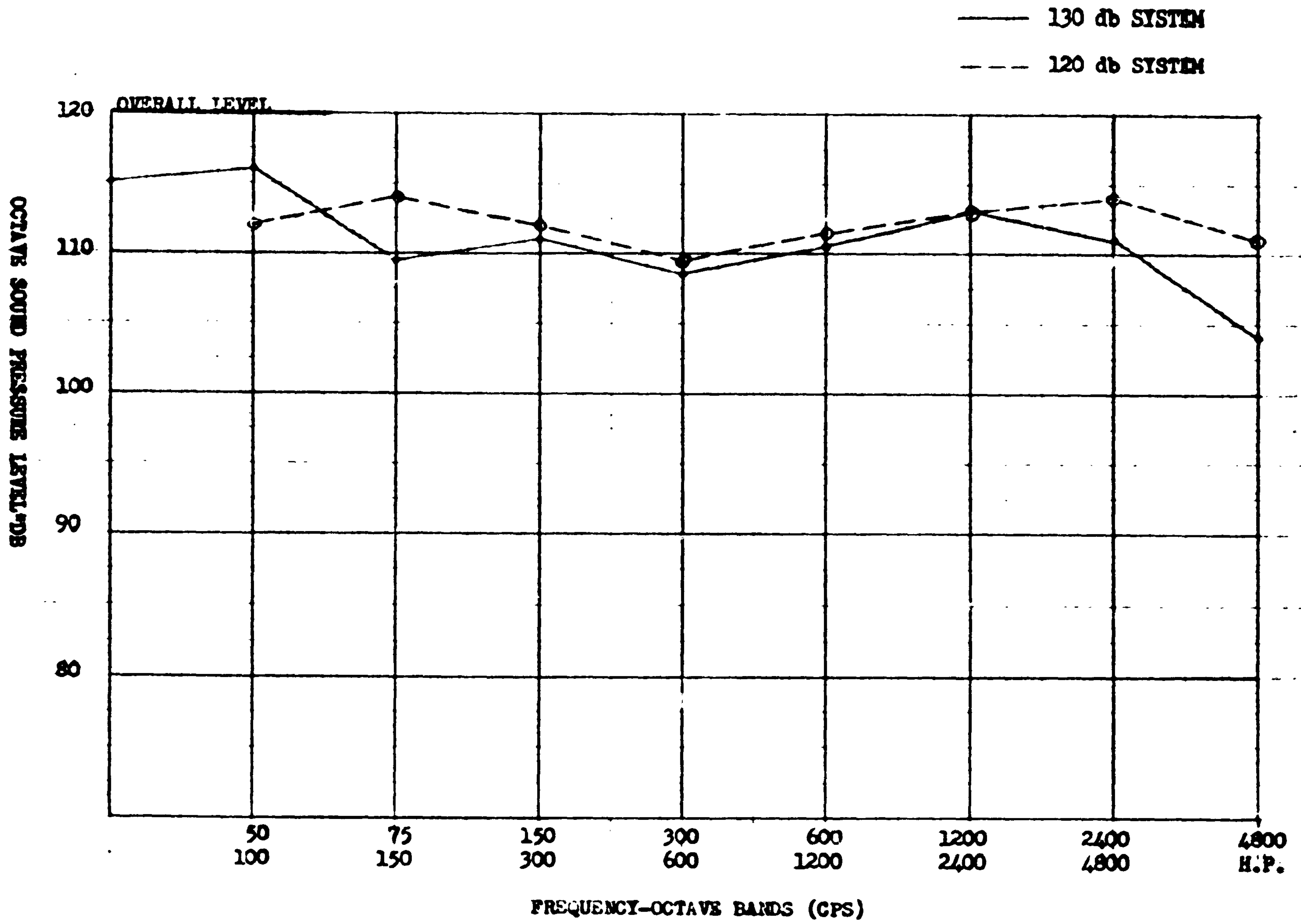
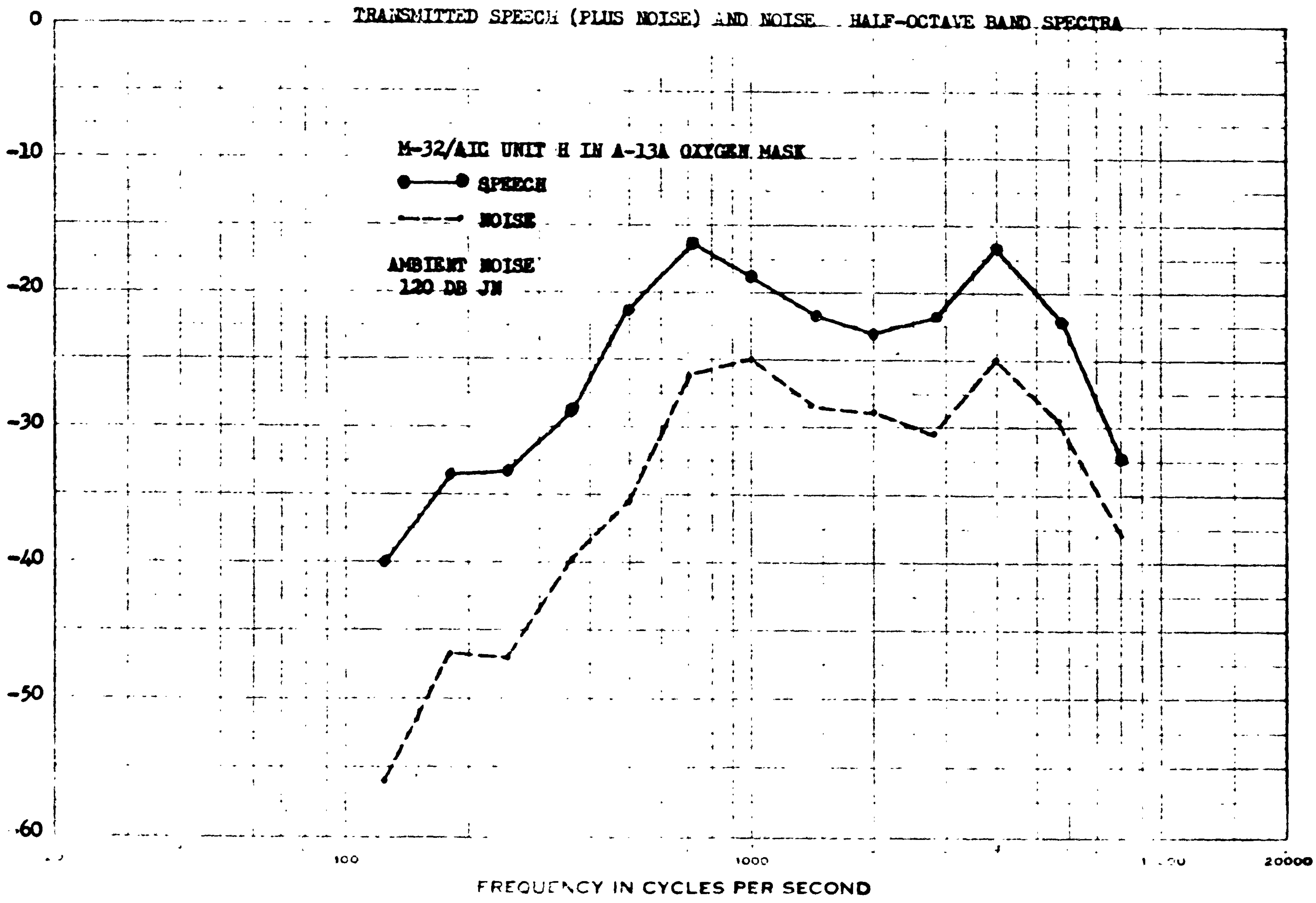


FIGURE 8

FIGURE A2-8



LEVEL - DB (RE 1 MILLIVOLT)
FIGURE A2-9

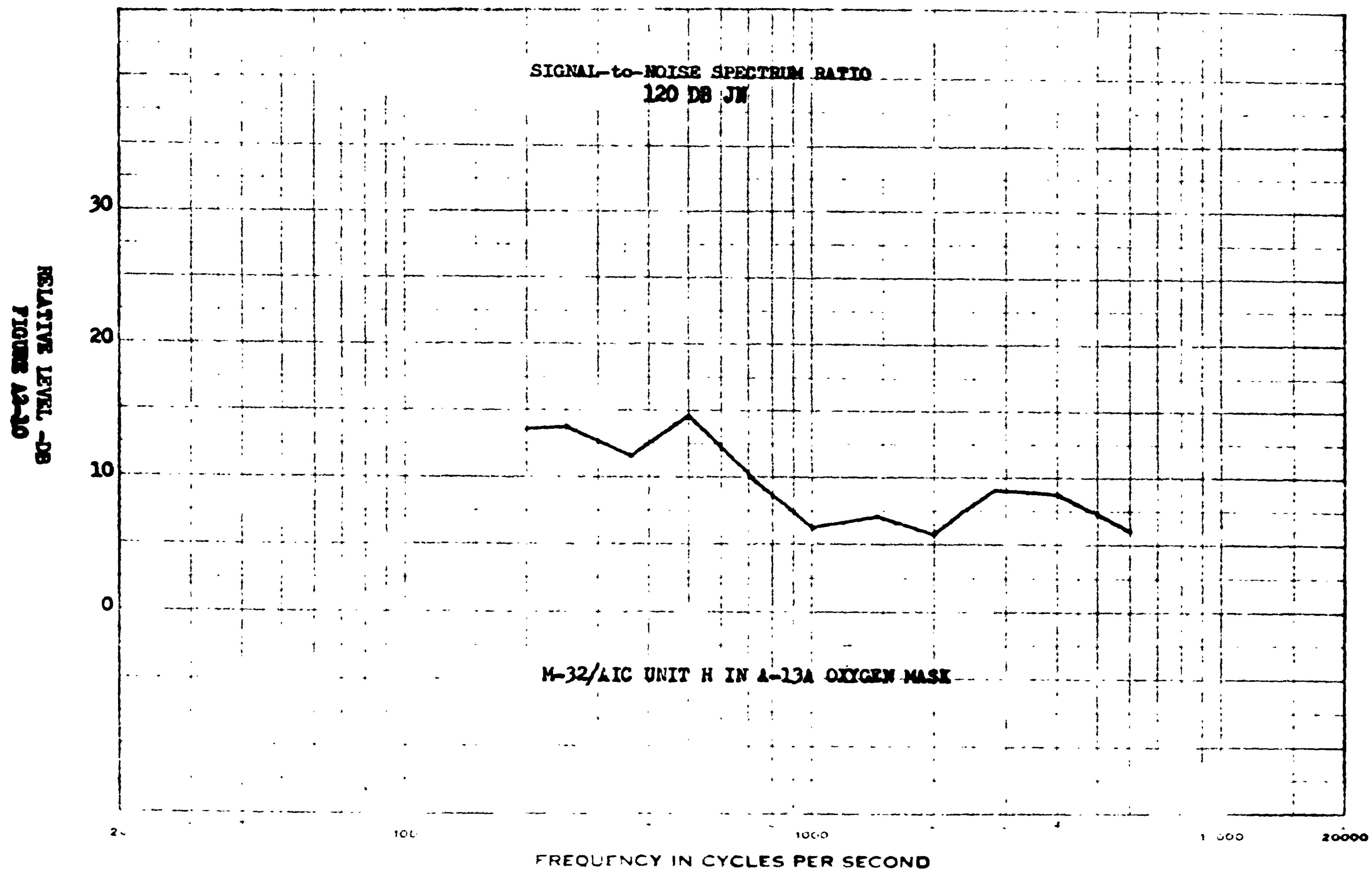


FIGURE 10

FIGURE A2-10

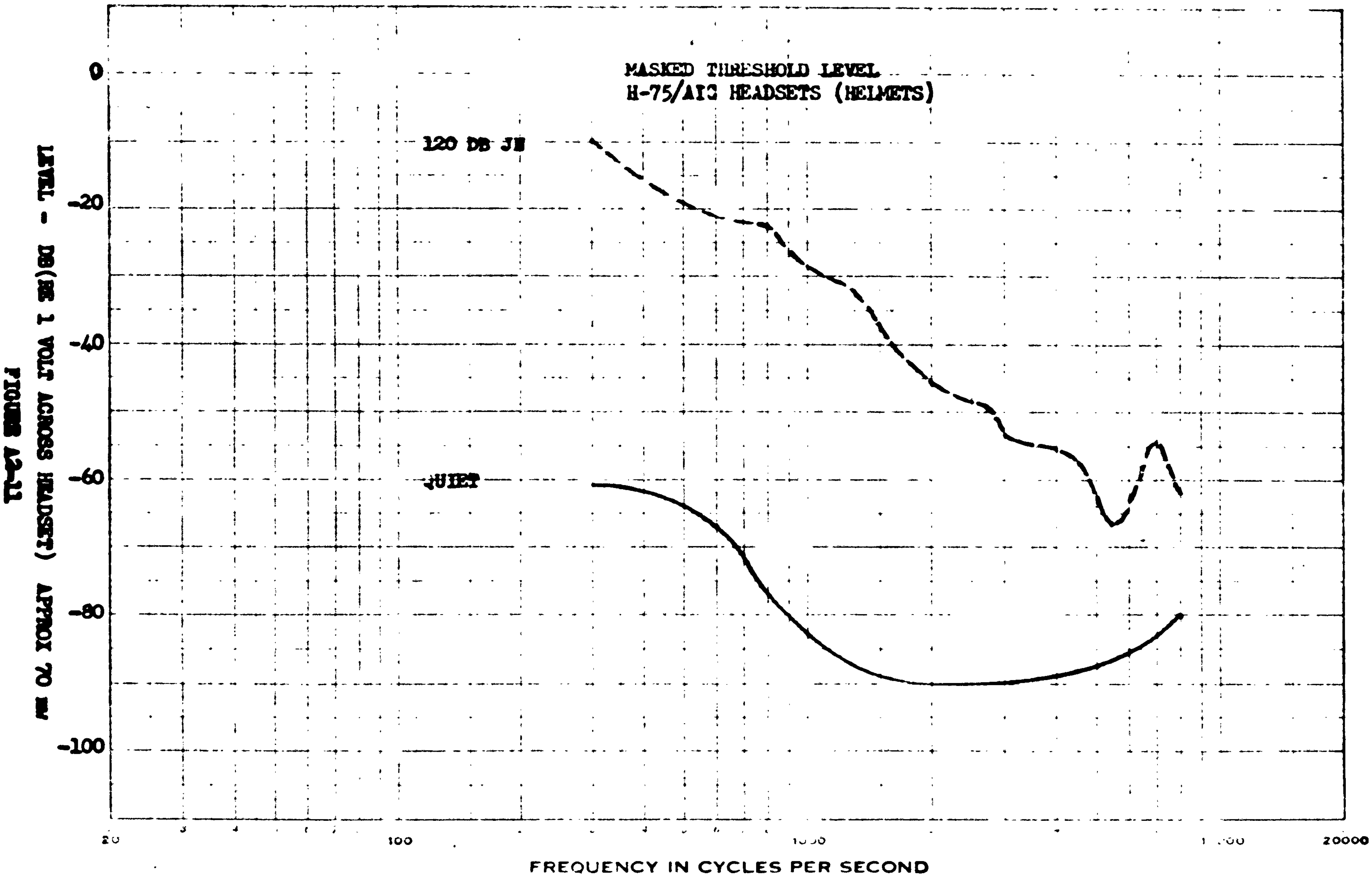
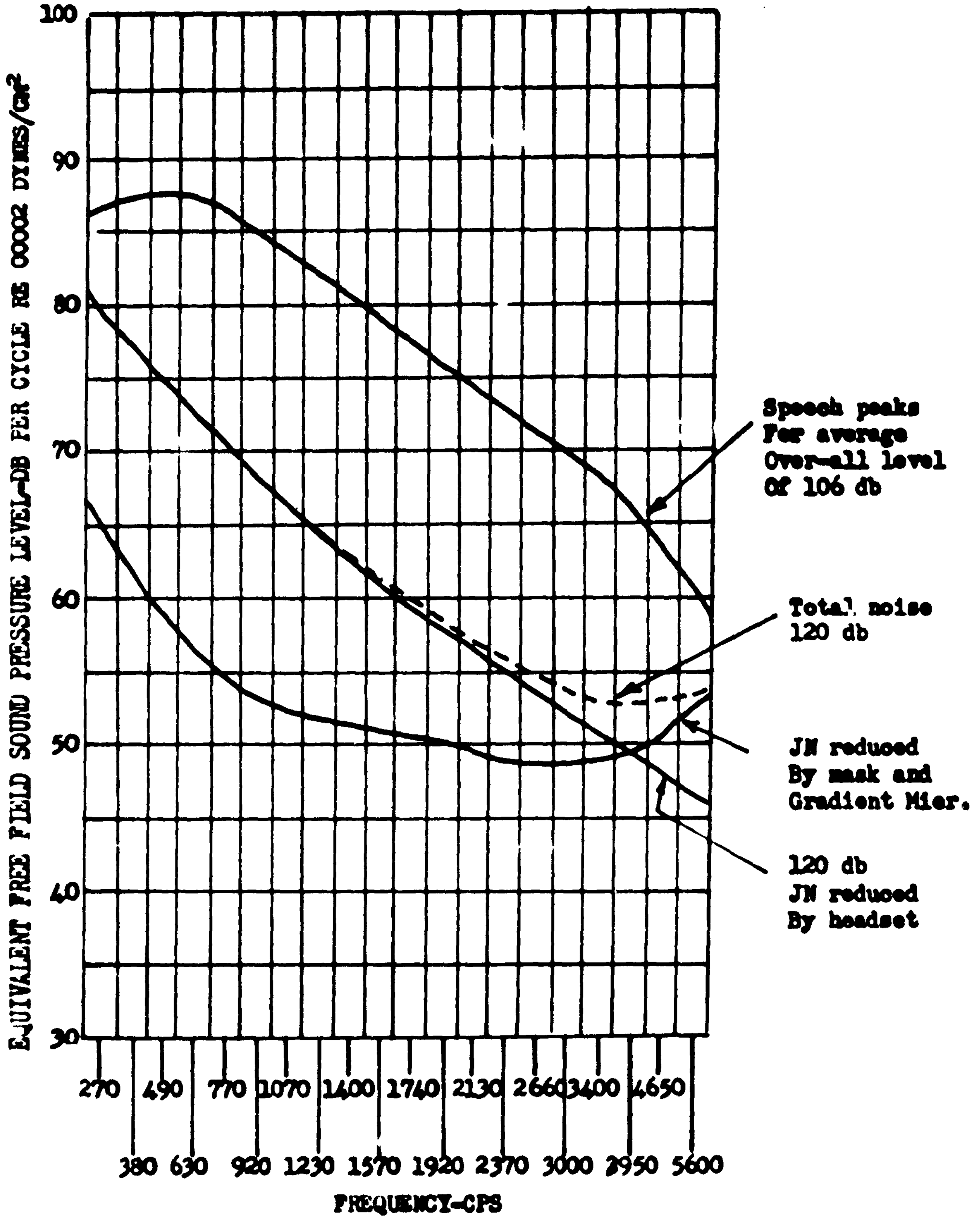


FIGURE A2-11

FIGURE 11



OXYGEN MASK GRADIENT MICROPHONE,
HEADSET SYSTEM PROBLEM, 120DB
JET NOISE

Figure A2-12

ARTICULATION SCORES

AN/AIC-10 IN 120-130 db

JET NOISE LEVELS

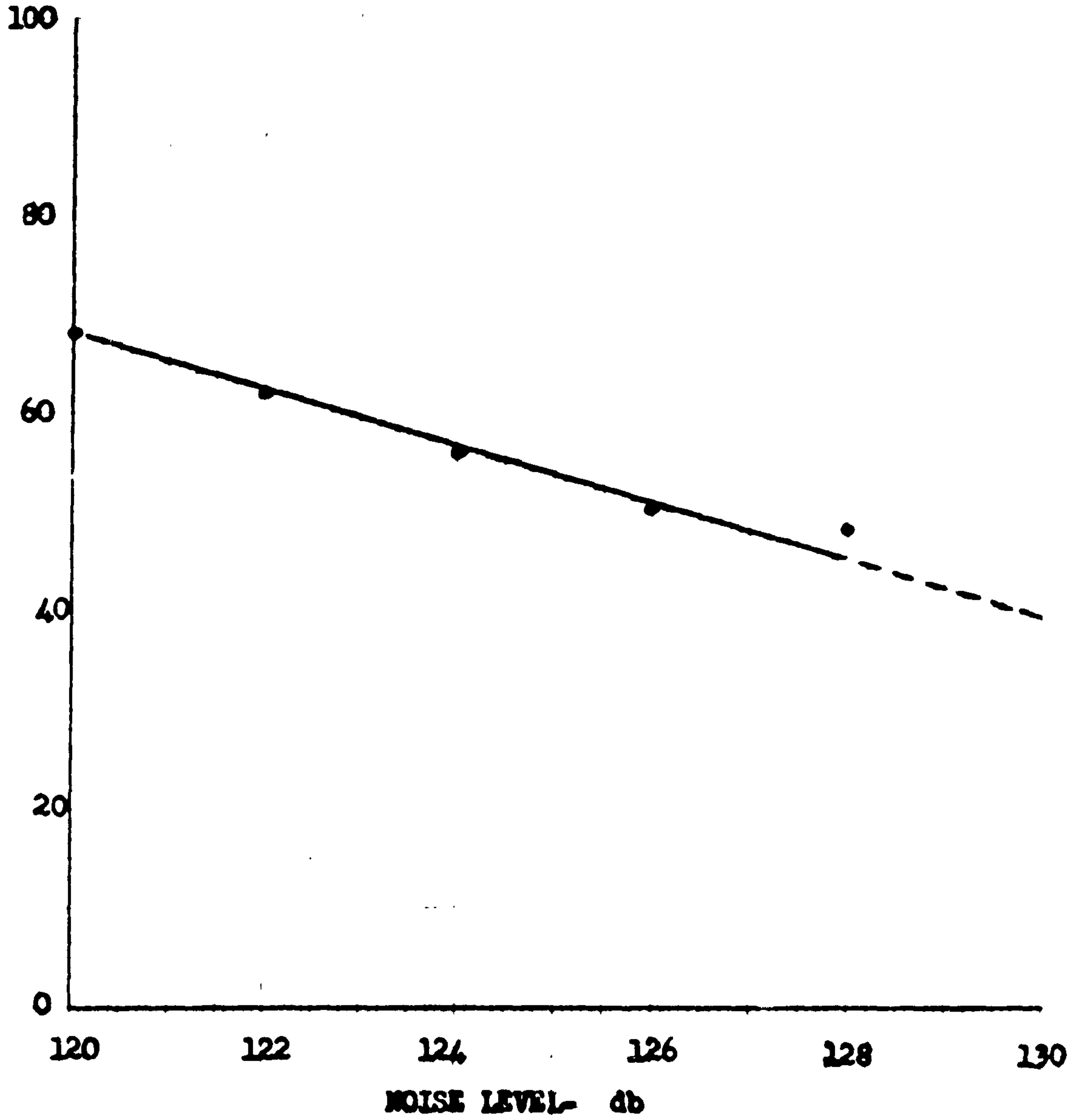


FIGURE A2-13

PPENDIX 3.0

STUDY OF TRANSDUCER TYPES

BY

W. B. SNOW
T. J. SCHULTZ

APPENDIX 3

STUDY OF TRANSDUCER TYPES

Appendix

3.0 STUDY OF TRANSDUCER TYPES BY W. B. SNOW

3.1 GENERAL CONSIDERATIONS OF SPEECH TRANSMISSION AND RECEPTION

3.1.1 SPEECH TO NOISE RATIO

In quiet surroundings, surprisingly low speech levels yield high intelligibility to those with normal hearing (Ref.1). People with hearing impairment may have difficulty under these conditions. When the listener is surrounded by noise, the speech levels must be raised until they exceed the noise levels in the frequency regions important to speech perception, namely 250 to 6000 cps for perfect intelligibility, or 400 to 4000 cps for good intelligibility. With the increased speech levels the person with impaired hearing will have less difficulty or none at all.

It is necessary to have only a moderate margin between speech and noise, provided this margin is obtained throughout the frequency range. This is demonstrated well in the calculations of RCA Progress Reports No. 1 and 2 (Ref. 2). However, with poor equipment, or even with good equipment mismatched to the application, it is possible to have speech levels in some frequency regions which greatly exceed the noise, while in other regions they do not. Such conditions result in high, annoying and unnecessary speech levels without good intelligibility.

Adequate hearing depends, then, upon the ratio of speech levels to noise levels, rather than upon a particular speech level. If the noise is high, it can be overcome by a high speech level. But if the noise at the ear can be reduced, adequate intelligibility can be obtained with lower and more acceptable speech levels.

3.1.2 NOISE REDUCTION

There are, of course, limits to the process of providing higher speech levels. It is because these limits are being reached in current situations that this study was undertaken. From the standpoint of the man, prolonged exposure to high levels can produce permanent hearing loss, in addition to the annoyance. Possibly this aspect of the system may be considered as a military hazard which must be accepted if necessary. The exposure is not continuous, and the hearing loss will be of a type which may affect the man's activities in quiet surroundings but will not interfere appreciably with his speech perception in the noise.

However, another physiological aspect of very high speech levels does have extremely important military significance. This is the fact that the ability to understand speech decreases at these levels, even though they can be tolerated. Consequently the efficiency of communication suffers. If both speech and noise levels are reduced, better communication results for the

same ratio of speech and noise. This adds up to the fact that there is a top limit for speech levels which can be used, and if voice communication is to be achieved the noise levels must somehow be maintained sufficiently below this limit to allow intelligibility. In this region any addition to the noise shielding pays an extra dividend in added intelligibility.

There is a distinct engineering advantage to noise reduction in addition to the factors pertaining only to hearing. A decrease in noise allows a corresponding decrease in the power required from the earphone or loudspeaker and its driving amplifier. At low and moderate noise levels this is not of much concern. The minimum physical size of components practicable from a construction standpoint will furnish all the power required. In the high noise fields in military aircraft this is by no means true. While the necessary powers do not represent a significant drain on the aircraft electrical system, they do necessitate equipment of size and weight which become serious problems. In addition, for instruments which satisfy adequate wearability standards, high acoustic power requirements pose other problems. The instruments may not be capable of great enough diaphragm movement, or may not be able to dissipate the heat adequately or safely.

From the standpoint of effective communication, therefore, it is extremely important to reduce the noise levels at the listener's ears (or head) to reasonable values. Conversely, of course, this process need not be carried past the point where the smallest equipment is more than adequate. However, noise shielding is so difficult to achieve in aircraft in the face of all the conflicting requirements of wearability, personnel mobility and weight, that there appears no likelihood of invading this latter region. It appears certain that all feasible noise reduction will aid development of the most effective communication system.

Noise reduction is important at the microphone end of the system as well as at the listening end. A talker can produce a definitely limited amount of sound (Ref. 3). If this sound is not sufficiently above the noise picked up by the microphone, the speech is irretrievably lost. Once the speech and noise become mixed in the microphone output, nothing can be done to improve the speech-to-noise ratio on a spectrum basis, i.e. in each individual frequency region over the important range. Consequently, the microphone end of the system also has an upper noise limit beyond which speech communication becomes impossible.

3.1.3 TRANSDUCTION - POWER AVAILABLE

The microphones and earphones or loudspeakers of a speech communication system are electroacoustic transducers - devices that convert electric quantities into acoustic quantities, or vice versa. In acoustic form these quantities are sound pressure, particle velocity and energy, while in electric form they are voltage, current and energy. For the transducers useful in this study there is always a mechanical structure interposed between the electric and acoustic ends of the transducer which makes the conversion and affects the results. Since there are many forms of transducers operating on a number of principles, there are many impedance variations in the instruments and the coupled electric circuits.

It is always true that the power or energy delivered by the transducer equals the power it absorbs minus the power lost in its own action, usually in the form of heat. This lost power may be anything from very low to very high, making the transducer have high or low inherent efficiency. However, power can be transferred only to a resistance, even though high currents, voltages, pressures or particle velocities may exist in associated reactances. If reactance is high, the actual power transfer may be determined more by mismatched impedance than by the inherent transducer efficiency.

The term "power available" has been worked out to simplify the situation. It is an extremely useful concept. When a generator or a load are not purely resistive, the power transfer varies with frequency because the impedances vary. The power available is the power which the generator would deliver to a load with resistance equal to the generator resistance - in other words, to a load which tunes the generator at the frequency of interest. This is the maximum power which can be drawn from the generator. The ratio of the actual power delivered to the power available is a measure of the reduction in transfer-effectiveness caused by the reactances of the circuit, which "use up" voltage without contributing power. Expressed in db, this ratio is frequently called the "power available response."

In speech communication systems it is not possible to tune the circuits because at least several octaves must be transmitted. For these systems, therefore, the power available concept has great usefulness. It gives a method of easily specifying an unambiguous circuit configuration which corresponds to practical operating conditions. Whenever actual energy transfer is the important consideration, it is a most convenient method of giving data.

Energy transfer is not always the necessary condition, as will appear later in the discussion of microphones and earphones. In these cases, other criteria apply. Particular applications of the use of both power available and other criteria will be given in following sections.

3.2 MICROPHONES

3.2.1 MICROPHONE TYPE DESCRIPTIONS

Electrodynamic

A diaphragm carries a cylindrical coil which moves back and forth in the radial magnetic field in a slot into which the coil fits. The voltage induced in the coil is the microphone output.

Electromagnetic

Motion of a diaphragm varies the magnetic reluctance of an airgap in the path of flux from a permanent magnet. Fixed coils surround the variable flux path and the voltage generated in these coils is the microphone output.

Pole-Price Magnetic: The magnet is at the center of the diaphragm. Flux passes through the whole diaphragm.

Ring Armature Magnetic: The magnet is a ring around the periphery of the diaphragm. Flux passes through only the outside edge of the latter.

Reed Armature Variable Reluctance: The magnetic circuit is a reed clamped at one end and projecting out over the magnet. The reed is driven by a link from the diaphragm.

Balanced Armature Variable Reluctance: Similar to the reed armature, except that the "reed" is supported at the center and both ends move in magnetized gaps.

Electrostatic

The diaphragm is very thin and forms one plate of a capacitor. As it is moved by sound pressure, the capacitance is varied.

Condenser Microphone: The capacitor has dc and polarizing voltage. Capacitance changes result in charging currents through a resistor, the resulting voltage across which is amplified.

Radio Frequency Capacitor Microphone: The capacitor forms part of a tuned circuit supplied with radio frequency, or part of an oscillator tuning circuit. The effects on tuning of the capacitance changes are detected to get the audio signal.

Piezoelectric and Electrostrictive

Materials which generate an electrical voltage between terminals fastened to opposite faces when subjected to stress.

Piezoelectric materials such as Rochelle Salt and ADP crystals have this property inherently. The titanate materials gain the effect when they are charged with a high voltage and retain the charge. In either case the impedance is essentially capacitive and relatively high.

Bimorph Type: The active material is made into a sandwich, similar to a bi-metal strip, which is clamped at one place and driven by the diaphragm at another to gain maximum output from the high stress induced by bending.

Expander Type: The material is driven directly by the diaphragm in a compression-expansion direction without bending.

Variable Resistance

All of these microphones operate upon the principle that an effect of the sound alters their resistance, changing the current in an external polarizing supply circuit. Polarization is ordinarily simple dc, but need not be.

Carbon: The diaphragm compresses a mass of carbon granules, and the changes in contact surface alter the resistance.

Semi-Conductor (Germanium) and Strain Gage: The diaphragm motions change the stress in the material which has the property of changing resistance with stress.

Thermistor and Hot Wire: These materials, which vary their resistance as a function of temperature, sense the temperature alterations resulting from the varying gas conditions produced by the sound wave.

Electron Tube: The grid of the tube is moved by the diaphragm motion, resulting in changes in plate current.

MISCELLANEOUS

Magnetostrictive: The diaphragm is connected to a magnetostrictive material, which varies its magnetic reluctances as a function of stress, so as to change the stress. The material carries magnetic flux and is surrounded by a coil. The flux changes due to the varying reluctance induce voltages in the coil. Impedance is mainly inductive and usually low.

3.2.2 SENSITIVITY RATINGS

The sensitivity of a microphone indicates its effectiveness in producing an electrical signal from the sound actuating it. The commonly used figure gives the generated open circuit voltage level in db vs 1 volt, when the actuating sound pressure is 1 microbar (1 dyne/cm^2). To be useful, this number must be coupled to another, which is the microphone impedance. A low impedance microphone with low open circuit voltage can deliver as much electrical power as a microphone with high voltage, but also high internal impedance.

In recognition of this effect RETMA rating number G_M was standardized. It is proportional to the power available from the microphone for a specified sound field, and thus gives a rating of the microphone's effectiveness as a converter of acoustic to electric energy. It includes the actual microphone losses and impedance mismatches, as well as the effects of the microphone acoustic reactance in reducing the absorption of the full acoustic power available in the sound field. The G_M rating is primarily useful where the generator is resistive, or at least has a relatively large resistive component. This is true of microphones operating on resistive or magnetic principles. For them a representative rating resistance may be chosen, and the G_M value is valid over the audio frequency range. They are characteristically of low impedance. Present simple transistor amplifiers have low input resistance. The low impedance microphones are well adapted to work with them, and the G_M rating shows the relative outputs that will be obtained.

Conversely, microphones depending on variations in electric charge are of high impedance, largely capacitively reactive. They have been developed because the vacuum tube has a high impedance input circuit, also capacitive, and requires only voltage to control its output. Consequently, power output from the microphone is not required, and frequency distortion does not result. In this case the G_M value has little practical significance. For the present, with simple circuitry, the high impedance microphones must continue to be used with a vacuum tube as an input stage. Since this can be a very small and rugged device operating at low voltage to feed a transistor amplifier, it is not a serious restriction on the use of high impedance microphones if they have other advantages. Develop-

ment of the transistor art may remove even this small complication in the future.

A valid indicator of microphone sensitivity for all microphones is the voltage which the instrument will deliver to a vacuum tube grid under practical operating conditions. The high impedance instruments can be directly coupled. Low impedance microphones are coupled through a transformer, which in this report is assumed to transform the microphone rating impedance to 100,000 ohms - a figure easily attained for the voice-frequency range. In this case, the open-circuit voltage of the microphone drives the vacuum tube. This gives a gain of 6 db in signal and 3 db in inherent signal-to-noise (Ref. 4) over the loaded condition envisaged in the G_M rating circuit. Such a rating is given in the tables of a following section, in addition to G_M.

3.2.3 SENSITIVITY AND RESPONSE CRITERIA

Most microphones were developed with quite different objectives than those characterizing this study. They were to be used under conditions of low noise and low signal level, so that high sensitivity was a prime necessity to reduce noise to a minimum in the reproduction. In addition, both for sound systems and measuring instruments, maximum uniformity of response vs. frequency over a wide frequency range was desired.

For the helmet microphone, conditions are different. The sound level will always be high because the pickup point is so close to the lips, and even at best the noise level will also be high. Consequently, high sensitivity is not of the same crucial importance as in other microphones. Even with relatively low sensitivity the necessary amplifier gain will be less than that required in ordinary systems. This should not be interpreted as saying that sensitivity is not important, because it is. The more output obtained, the lower the gain that must be supplied by the amplifier, and the greater the margin over electrical pickup noise in the system. The true imterpretation is that microphone types or sizes which could not be considered in broadcast picur, for example, may be applicable to the helmet situation if they have compensating advantages. Examples of these advantages are size and weight reduction, resistance to rugged environmental conditions, ease of manufacture, constancy of output with altitude change, and noise reduction possibilities.

Similar considerations apply to frequency response. In the mask microphone a falling characteristic at low frequencies is desirable, because of the pressure buildup inside a helmet or mask which occurs in this frequency range (Ref. 5). If such a characteristic is not inherent in the structure, it will have to be added to the electrical circuit.

Frequency response outside the intelligence-bearing range is actually undesirable, for it only brings in added noise. This is a distinct difference from the criteria for sound system microphones used with music or speech in quiet, where naturalness is of much greater importance. Thus, intuitive ideas on microphones based upon much past experience have to be adjusted to these new criteria, which are akin to hearing aid criteria.

A requirement peculiar to aircraft systems is the desirability of maintaining constant response characteristics at all altitudes. This is one of the most serious problems of design, because microphone structures inherently contain acoustic networks and usually employ additional acoustic networks for response control. The characteristics of such networks vary with air density. Far from being an academic question, the specification of operating altitudes and tolerable variations in response over the necessary altitude range is one of the crucial decisions in transducer design.

3.2.4 ELECTRICAL CONSIDERATIONS

Low impedance microphones can employ long cables without any effect upon their sensitivity or characteristics (long cables do invite pickup troubles, however). High impedance instruments may be severely limited as to cable length. If the cable capacitance approaches or exceeds that of the microphone, the sensitivity will be reduced by the voltage-dividing action of the microphone and cable capacitances, but the frequency response will not be affected. High impedance circuits require careful shielding against electrostatic pickup. They are susceptible to effects caused by moisture-induced leakage resistance. The high impedance instruments give a high voltage output to a high-input-impedance amplifier, obviating the transformer required for equivalent voltage from a low impedance instrument. The relative characteristics are well brought out in the tables below.

A lower limit to microphone sensitivity is always set by inherent electrical noise, even in the complete absence of any interference pickup. For self-generating types this noise arises in the resistance of the microphone. For the variable resistance type (e.g. carbon or germanium) additional noise is generated by the passage of the required polarizing current.

Vacuum tubes also have an inherent noise 'floor' in the grid circuit, which is in the range of that generated by a resistance of 1000 to 10,000 ohms. With low impedance microphones whose resistance is transformed to about 100,000 ohms facing the grid, the tube noise contribution is negligible. But with the capacitive types, the tube noise and grid resistor noise set the limit. Current simple transistor amplifiers have

input resistances in the range of 1000 to 10,000 ohms. They are frequently coupled by a matching transformer, so that they load the microphone, reducing the signal-to-noise ratio. Transistors also have a certain amount of noise in addition to the pure resistance noise. The combination of these effects yields a somewhat lower inherent signal-to-noise ratio for the transistors.

In the helmet microphone this inherent noise is of negligible importance with any probable design, because the inherent signal-to-noise ratios for both vacuum tube and transistor amplifiers are so much greater than those obtainable because of ambient noise in the helmet that inherent amplifier noise is unlikely to be a problem.

3.2.5 NOISE REDUCTION

The importance of noise reduction at the microphone has been pointed out. This is partially accomplished by the shielding of the helmet and placement of the microphone close to the lips. An additional expedient is the use of the noise-cancelling type of instrument. This type admits sound to both sides of the diaphragm, and, particularly at low frequencies, yields a low noise output because the noise can build up little pressure difference across the diaphragm. When such a unit is used very close to the lips, speech sounds do not reach both sides of the diaphragm equally well, and higher output results.

3.3 SPEECH RECEPTION

3.3.1 INTRODUCTION

Transducers for reproducing sound have been built, or proposed, using a wide range of principles just as in the case of microphones. For these transducers, as for microphones, the nature of the helmet intercommunication problem imposes some requirements differing from those usually applied. The requirements for frequency range limited to the voice band but changing as little as possible with altitude, resistance to difficult environmental conditions, minimum size and weight all apply. It has been found that wearability problems are even more difficult than for microphones.

At this reproducing end of the system, efficiency is more important than at the microphone end, where a single small vacuum tube or transistor can add 30 db or more of amplification. Here the tubes or transistors are supplying power, and every 3 db increase in required amplifier output necessitates doubling the number of power tubes or resistors, or going to larger units with double power capacity. This is not the only limitation. Even if

amplifier power can be conveniently raised, the transducer structure may not be able to accept the increase without overheating or objectionable distortion. Therefore, the transducer should convert as much of the electrical power available as possible into acoustic power or sound pressure, subject to the many other requirements on its structure or performance.

Assuming air conduction, the only thing that counts is producing sound pressure variations on the ear drum at the bottom of the ear canal, because the drum is a pressure-sensing pickup. Outside of the canal, any sound is wasted which does not reach essentially the canal opening and create sound pressures in it.

On the other hand, the entrance to the canal does not stay fixed, for the man must move about. The source must go along with the ears, or a sound field of sufficient strength must be provided at all places where the ears will be. Thus, possible solutions to the problem run from a tiny insert earphone to loudspeakers filling the cockpit with sound. The characteristics of the various methods are discussed in the following sections.

3.3.2 EARPHONES

3.3.2.1 SENSITIVITY

An earphone does not radiate sound as a loudspeaker does. It works into a closed cavity terminated by the eardrum and its job is to create pressure changes in this cavity. A small unit under these conditions can work effectively at low frequencies where it would be completely ineffective as an open-air loudspeaker. In essence, it is pushing against a spring and not supplying much power.

An earphone is essentially a piston moving in and out of one wall of a closed cavity. If the cavity is small, a given motion of the piston (earphone diaphragm) will cause greater pressure changes than in a large cavity. A small insert phone fitted into the ear canal can cause a high sound level with relatively low drive power because the enclosed volume of air to be compressed is 2 cc or less. A regular-sized earphone with a small cap pressed against the ear encloses a volume of about 6 cc and also produces relatively high sound levels for the power input. Unfortunately, such an arrangement has very poor wearability characteristics. For short periods, as in telephony, it is satisfactory; but for long periods it is not. Padded ear caps aid wearability but increase volume, while circumaural designs which completely enclose the pinna and contact only the head develop cavities of 30 to 70 cc and [illegible] losses in sound level of 10 to 15 db for moving coil earphones. This is the price paid for wearability.

The problem is not that simple, however. In a noise environment the speech intelligibility depends upon the ratio of signal-to-noise, not upon either alone. It is just as effective to reduce the noise 10 db as to increase the signal 10 db. Unless there is a tight seal between the earphone and the ear or head, the leakage reduces the sound level generated by the phone, particularly at low frequencies. Also, the leakage allows noise to enter from outside. Small flat earcaps do not give a good seal, and therefore do not exclude noise well. Some of the larger-volume cushions, and particularly the latest designs of fluid-seal, hard-shell circumaural cushions, have high noise exclusion. In this type of earphone the greater volume increases the noise reduction even though it decreases the speech levels. Consequently, the disadvantage of the lower signal level is partially or even completely offset by the lower noise level. At the same time, wearability is increased.

The insert phone system not only gives small volume, but good noise exclusion. It is therefore acoustically very effective. Wearability is the chief factor which will determine its practicability.

3.3.2.2 RESPONSE

The structure of an earphone is such that it would have resonances and irregular frequency response unless measures to smooth its characteristics were employed. This is done by building into the structure acoustic cavities, ducts, openings and damping screens which control the mechanical properties by motion and friction in the air. But the proper action of these devices depends upon the air density, so that at high altitudes the characteristics change. In addition, the sound radiated is reduced at high altitude.

To correct this effect the acoustic networks in the earphones of the AN/AIC-10 system are over-corrected at sea level pressure. The diaphragm works against a cavity so small, and therefore stiff, that the motion is reduced considerably. As the density decreases, this loading effect decreases, permitting the diaphragm to move farther and to generate approximately the same levels at altitude as at sea level. This choice of construction is based upon the premise that the system must have sufficient sound level to work at altitude, and that it is an actual advantage to hold down the earphone efficiency at lower altitudes, thus to maintain constant performance without circuit gain adjustment. This emphasizes the necessity of examining earphone response at the highest altitude where full performance must be provided, as well as at ground level. As in the case of microphones, the most ideal design for ordinary uses may not even be usable for the intercommunication system.

These statements are based upon results with the types of earphones that have been used in the past, and do not apply entirely to the electrostatic earphone. The latter type

has been given special study for this report, and Appendix details the results, both for earphones and loudspeakers.

ELECTRICAL DRIVE

3.3.2.3 ELECTRICAL DRIVE

Both vacuum tube and transistor amplifiers have resistive output impedance. There is no difficulty in driving an electro-dynamic earphone from such an amplifier, since those phones have essentially resistive impedance and the current remains essentially constant at all frequencies. This also applies nearly enough to the electro-magnetic types which are only mildly reactive.

The capacitive transducers cannot be driven by a resistive generator over a wide frequency range in a manner that truly exploits their efficiency. The resistance component of impedance of these units is usually very small and if they are tuned the efficiency is high. For wide-band operation the generator impedance must be selected arbitrarily to equal the absolute transducer impedance at some frequency in the band, and the reactance determines the voltage which will be applied to the unit at other frequencies. Thus it is necessary to choose a driving resistance which gives the best characteristic for the job in hand. The power available concept makes it possible to state the responses of all types of earphones on a comparable basis.

3.3.3 LOUDSPEAKERS

3.3.3.1 RADIATION AND DIRECTIVITY

A loudspeaker differs from an earphone in that it radiates sound energy into space. To do this effectively, it must have an "outlet" size comparable to the wave length of the sound, i.e. greater than 1/4 wave length. For direct radiators this means the actual size of the diaphragm; in horn loudspeakers it means the size of the horn mouth, where the diaphragm can be small if the horn is long enough. Because the speech frequency range covers a band in which the lowest frequency is about 1/20th of the highest, the outlet size will be several times the highest frequency wave length. This introduces the effect called the directivity of the loudspeaker; the directivity increases as frequency increases. For communication purposes it would be desirable to project sound only to useful areas without wasting it elsewhere. Practical loudspeakers are not this directive. At the low frequency end of the range they are essentially non-directive and distribute sound effectively in all directions. It is not always appreciated that this applies to horns as well as to direct radiators <u>of the same size</u>. (Ref. 6). As frequency increases and the dimensions of the loudspeaker become larger compared to sound wave length, the sound is radiated into a narrower angular zone, and a confined beam results.

Many refinements in design of horns and direct radiators are used to control this effect - usually to widen an otherwise too narrow beam at high frequencies. As noted above, a horn cannot concentrate sound at frequencies where the wave length is relatively long, but at higher frequencies where a direct radiator has a narrow beam, a properly designed horn distributes the sound with a moderate beam width which remains relatively constant with frequency. The effects of directivity on speech perception are illustrated by data in Baldwin report (Ref. 7) describing tests on the AN/AIC-10 direct radiator loudspeaker. Observers seated off-axis obtained significantly poorer articulation scores in high noise.

3.3.3.2 EFFICIENCY

High efficiency in a loudspeaker is of even greater importance for this application than for earphones, because higher powers are involved. In any loudspeaker the portion of input energy which is radiated as sound is the only portion which is useful except very close to the unit. Power can only be supplied to a resistance. The sound radiation is represented on the electrical side of the transducer as a part of the total resistance - the rest constituting losses which merely serve to heat the transducer. The impedance of a perfect loudspeaker would be a pure resistance, all of it representing radiation. Some physical transducers of the electro-dynamic or electro-magnetic type approach this over a reasonably wide frequency range, but even they have maximum efficiencies of only 25 to 40%. As in the case of earphones, these types are easy to drive from resistive amplifiers. Capacitive types have low losses, but cannot be driven to take full advantage of their inherent efficiency. Although the power available must be made large, the transducer will not accept much of it. It does an efficient job of transducing from electrical to acoustic form what it does accept.

It should be pointed out that exactly the same fundamental situation holds for magnetic devices where the reactance is inductive rather than capacitive. In general, however, magnetic transducers have a relatively high ratio of resistance to reactance and the impedance does not change a large amount over the audio frequency band. This means that they do accept the power, but heat themselves up with most of it rather than producing sound from it.

No matter how light the motor elements of most loudspeakers are made, they are massive compared to the air into which they are supposed to radiate sound. They can usually generate considerable force, but a limited amplitude of motion. They are therefore coupled to diaphragms so that a small element in the motor will move a much large volume of air and will thus be loaded by a force more nearly comparable to that which it can generate. The diaphragm may radiate directly, in which case it is made relatively large. An acoustically more efficient

method is to couple the diaphragm to a horn; then it is made of medium size, and the "trapped" air in the horn transforms the radiated sound pressure at the mouth to a much higher pressure at the throat against which the diaphragm can push.

Contrasted to the compact motor-diaphragm construction is the electrostatic loudspeaker now coming into commercial use. In this case the motor and diaphragm elements are identical. A membrane is acted upon electrically and in turn radiates the sound into the air. It is so thin in relation to its length and breadth that its mass is almost negligible compared to that of the air it drives. It therefore has very high inherent efficiency, which partially makes up for its high capacitive impedance. For high output it must have high electric field strength, both for bias and signal, and the ultimate limitation is the dielectric strength of the insulating material on the stationary electrode. This requires high voltages, but they are supplied from extremely high impedance and are not hazardous. The characteristics of this class of transducer has appeared so interesting that a special study was made, reported in Appendix 3.4 . See also Ref. 8.

The loudspeaker size is determined by the frequency range over which high efficiency is required. It was pointed out that to radiate sound effectively the diaphragm or horn mouth must have dimensions comparable to the wavelength. Consequently, the lower limit of the necessary frequency range will dictate the minimum size of the loudspeaker.

An important aspect of the loudspeaker technique is that the best of present instruments have efficiencies of 25 to 40% over a considerable range. This is only 6 to 4 db below a perfect loudspeaker. The effectiveness of loudspeaker communication, therefore, must be determined primarily with units now attainable, because large increases in efficiency are not possible. There is, however, plenty of room for development of small, compact units equalling the present best in efficiency.

3.3.3.3 USE OF LOUDSPEAKERS

Loudspeakers might be used in this aircraft intercommunication problem for two distinct purposes. One would be as a substitute for earphones for pilots in a fixed location, to eliminate the necessity of wearing any tight-fitting ear covering. The other would be the more usual situation where the loudspeaker is used to provide speech over an appreciable area to several persons, or to allow mobility to one person. Both of these systems fall in the class of brute force methods since the loudspeaker sound competes directly with the full interfering noise. The system must create speech sound levels sufficiently greater than the noise levels so that the speech can be understood. If noise protection gear is provided, it attenuates speech and noise alike. Consequently, this type of operation trades mobility of the personnel and freedom from the necessity of wearing any hearing apparatus and connecting cords, for the greater bulk and weight of equipment which is mounted on the frame of the aircraft.

1.4 COVERING AN EXTENDED AREA

If the listener were out-of-doors, the on-axis response (or the actual response at his angle) would be the only consideration determining speech recognition. Sound radiated in other directions would have no effect. The loudspeaker characteristics published by manufacturers are almost always measured under conditions simulating out-of-doors listening. The sound received in this way is known as direct sound, since it travels directly from loudspeaker to listener. It would not matter to an on-axis listener if the loudspeaker were directive. But if the characteristic were adjusted to give the best speech intelligibility in noise for on-axis listening with minimum power input, then a listener off-axis would receive about the same low-frequency level but reduced high frequency level with consequent reduced intelligibility.

In the aircraft the loudspeaker will be enclosed by a chamber which has walls that will absorb some sound, but will reflect a large portion of the energy. Under these conditions the effects of directivity are partly overcome. The sound bounces around in the space and the sound level in the cabin will build up to such a value that the sound energy absorbed by the surroundings just equals that emitted by the loudspeaker. Close to the loudspeaker the direct sound predominates, and a listener there will receive about the same sound as the listener out-of-doors. At a distance from the loudspeaker the sound consists mostly of the bouncing or reverberant energy, as it is called. This sound level is approximately the same throughout the space.

The reverberant level, consisting of the reflections from all directions, measures the total power radiated by the loudspeaker, modified by the absorption of the surroundings. The loudspeaker radiates relatively the same energy in all directions at low frequencies, but confines the energy to a narrower beam at high frequencies. Out-of-doors the widespread energy was lost; in the cabin it is collected. Therefore, the speech characteristi of the reverberant sound will differ from that of the closeup sound. The low frequencies will be accentuated, both because of the collection of more sound from the loudspeaker and because, in general, high frequency energy is more readily absorbed than low frequency energy by aircraft cabin surroundings.

It can be seen that enclosing the loudspeaker in a cabin makes use of its total energy output, much of which would be wasted out-of-doors. It might seem, then, that the cabin should be made as sound-reflective as possible to build up a high reproduced speech level. But this is not true, for two reasons. If the cabin is reflective, the level of ambient noise will build up as well as the speech level; nothing is gained in ratio, and

the absolute levels are higher. Secondly, the energy which builds up in the enclosure continues to bounce around after the speech syllable causing it has ceased, gradually decaying in level until it has been completely absorbed. If this process takes too long, old sounds interfere with perception of new sounds, and the gain in higher level is more than offset by the loss due to interference. It is probably true that any amount of sound absorption which will be tolerated in military aircraft from a weight standpoint will be an advantage to communication.

Certain conclusions can be drawn from this discussion. For this case of extended coverage, the loudspeaker should distribute sound at all frequencies over a wide angle and should maintain good total sound output efficiency to as high frequencies as possible. If it is necessary to tailor the response to a noise characteristic, this should be done for the reverberant field. Those listeners close to the loudspeaker may then not receive the optimum relative signal characteristic, but since they will be closer they will receive higher speech levels and will actually have better speech perception than those farther away. To obtain these effects in practice will require ingenious combinations of units; they will not be attained by a routine approach.

3.3.3.5 COVERAGE FOR SINGLE INDIVIDUAL

If the loudspeaker is used for a single pilot in a fixed location, it has to cover only those locations where he can put his head. The goal here would be to confine the sound to this particular space volume. This would be facilitated by a large directive radiator which would in effect always place him in the direct sound field. An alternate solution would be a system employing a group of small radiators spaced around the volume, or some form of "strip" radiator along the main path of movement. The effort here is to concentrate the sound in the necessary space and distribute as little as possible anywhere else. Probably maximum attainable directivity will be desired.

3.3.3.6 LOUDSPEAKER IN HELMET

An intermediate system is that in which a loudspeaker is used to reproduce the speech inside a helmet. Here the sound is confined to a small space which moves with the man and hence the system is more efficient. A loudspeaker operating into a closed helmet is in a position intermediate between the earphone and the normal loudspeaker. The unit will be working under unusual acoustical conditions, but its function will remain to produce the proper sound pressures at the ear in the particular helmet. At low frequencies it essentially creates only pressure changes because the wave length is long compared to the free dimensions of the helmet. At high frequencies it works more as

a normal loudspeaker, especially if acoustic absorption is used inside the helmet. While the volume is much greater than that under an earcap, it is still relatively small and a loudspeaker will build up in it a much higher sound pressure than it can produce in space or even in an enclosed cockpit. In this case, too, the ambient noise is reduced by the helmet so that the loudspeaker does not have to produce as great actual speech levels to override the noise as an external loudspeaker must produce outside the helmet. These two gains result in the greatly increased efficiency of this system over the external loudspeaker (Ref. 9).

The helmet volume is larger than earphone cushion volume, and the earphone cushions give additional noise shielding. It can probably be predicted that this type of a driver will outweigh headphone units; but the possibility exists that it will be worth it because of a more favorable mounting position and increased wearability time.

3.3.3.7 ACOUSTIC FEEDBACK OR SINGING

The loudspeaker systems are susceptible to acoustic feedback or singing if sound from a loudspeaker has a chance to enter a microphone connected to the same system. If the sound from the loudspeaker becomes greater than the speech level actually "recognized" by the microphone, the system will sing. A noise-cancelling microphone treats the loudspeaker sound as noise and gives the same reduction in singing susceptibility as it does to noise pickup. This is offset partially, at least, by the fact that the loudspeaker sound must be greater than the noise <u>at the listener</u> to be intelligible. Singing will have to be carefully considered in relation to loudspeaker systems and the side-tone level.

In this regard, it should be noted that the earphone systems, too, have the possibility of singing if sufficient side tone is furnished. In them, however, the attenuation of the ear cushion is in the singing path when the phones are on the ears; and if they are retracted the sensitivity falls because of the large volume of the helmet. It is not uncommon, however, for a hearing aid to sing when worn by a very deaf person who needs high amplification.

Appendix

3.4 The Electrostatic Transducer for Use in Cockpit Communications by T. J. Schultz

3.4.1 Introduction

Because electrostatic transducers have only recently become practical to design and produce commercially, their properties are relatively unfamiliar to engineers. This appendix will set out briefly the considerations and assumptions under which the figure in Chart 1 were calculated and will call attention to ways in which the behavior of the electrostatic units (ESU) differs strikingly from that of conventional transducers.

The form of the discussion will be as follows: first, expressions will be given for the sound generated by a transducer in terms of its geometry and the diaphragm velocity; then the velocity will be given in terms of the electrical, mechanical and acoustical characteristics of the transducer. By combining these equations with other appropriate relations, all of the relevant data can be obtained, and further interesting information deduced.

3.4.2 The Generated Pressure

It has been assumed in the case of the loudspeakers that they were plane surfaces radiating from an infinite baffle into a free field. For this condition, the axial pressure is given[10] (for frequencies above a limit to be discussed in paragraph 3.4.17) by:

$$p_{ax} = \frac{\rho f S v}{r} \qquad (N/m^2) \qquad (1)$$

where ρ is the density of the medium

f is the frequency (c/s)

S is the effective membrane area (m^2)

v is the average membrane velocity (m/s)

r is the distance between the loudspeaker and the point where p_{ax} is measured (m)

All entries in the loudspeaker chart were calculated for this condition. For the use of the ear inserts and the elements housed with helmets, it was assumed that the volume is sufficiently small that, to a useful approximation, no radiation occurs* in the frequency range of interest. Under these circumstances, the signal pressure is:[6]

$$p = v\left(\frac{Z_m'}{S}\right) = v \; \frac{S}{j2\pi f C_a} \qquad (N/m^2) \qquad (2)$$

*i.e. the wavelength λ is large compared to a typical dimension, and pressure changes are communicated instantly throughout the volume.

where p, v, s and f have the same significance as before, and C_a is the effective acoustic compliance* of the cavity into which the transducer communicates.

3.4.3 Dependence on Membrane Velocity Only

In comparing the output of an ESU with that from other types of transducers, all of the variable of eqs. (1) and (2) were considered fixed except v, the average velocity of the diaphragm or membrane. In each case, this velocity was assumed to depend upon some (constant) driving force (either magnetic or electrostatic) and upon the impedance associated with the diaphragm. The larger the impedance, the smaller the velocity of the membrane, and hence the smaller the pressure radiated from it. When we have calculated the membrane velocity, this can be put into eqs. (1) and (2) to find the pressure. Numerical values for ESU voltages and dimensions are, of course, subject to a wide range of design choices, but for these calculations, values were chosen which are typical of units now in production.

3.4.4 The Membrane Velocity

The discussion of this report is concerned solely with the ESU in "balanced - push - pull" operation because of its overwhelming advantages over the single-ended type. The push - pull ESU comprises two mobile conducting membranes, symmetrically supported between two fixed, acoustically transparent electrodes. A DC bias voltage is applied between the membrane (positive) and both stationary electrodes (negative) through a very high resistance, thus establishing a constant charge +q on the membrane. The signal voltage is applied to the two stationary electrodes to create a field between them which varies with the signal, both in magnitude and polarity, and in which the charge on the membrane is forced to move. A schematic diagram appears in Figure 1.**

The electro-mechano-acoustical equations for such a system are:

$$f_A = -Z_r v = Ti + Z_m v \tag{3a}$$

$$e = Z_e i + Tv \tag{3b}$$

where e is the signal voltage (volts)
i is the signal current (amperes)
v is the average membrane velocity (m/s)
Z_e is the electrical impedance of the transducer (electrical ohms)
Zr represents the radiation impedance (mechanical ohms)
Z_m is the mechanical impedance of the transducer (mech. ohms)
T is the electromechanical transduction coefficient

$$T = \frac{2\phi}{j\omega C_0} = \frac{2E_0}{j\omega d} = \phi Z_e \quad ; \quad \phi = \frac{C_0 E_0}{d}$$

* $C_a = \frac{V_0}{\rho c^2}$ $\left(\frac{m^5}{N}\right)$ where V_0 is the cavity volume (M^3). ρ is as above and c is the velocity of sound (m/s).

** We will discuss operation at the fundamental frequency only. An extensive discussion of harmonic distortion in such a system is given in Hunt's Electroacoustics, Wiley, (1954), pp 208-211.

The solution of eqs (3) for the symmetrical (and quite practical) case in which $d_a = d_b = d$, $C_a = C_b = C_o$, $q_a = q_b = q$ and

$$R_a = R_b = L_a = L_b = L_o = 0$$

so that $Z_e = \frac{2}{j\omega C_o}$ is:

$$v = \frac{-\frac{E_o e E_o}{d^2}}{\frac{Z_m}{S} + \frac{Z_r}{S} - \frac{1}{j\omega}\left(\frac{2E_o^2 \varepsilon_o}{d}\right)} = -\frac{2\varepsilon_o}{\left(\frac{Z_m'}{S}\right)}\left(\frac{E_o}{d}\right)\left(\frac{e_m}{2d}\right) ; \ (m/s) \quad (4)$$

where

E_o is the bias voltage (volts);
ε_o is the dialectric constant of the medium between the electrodes (farads/m);
d is the separation between the membrane and either stationary electrode (m),

$Z_m = j\omega l_m + r_m + \frac{1}{j\omega c_m}$ is the mechanical impedance of the membrane whose mass (K_g), resistance ($\frac{N-s}{m}$) and compliance (m/N) are l_m, r_m and c_m (mech. ohms = $\frac{N-s}{m}$);

$\frac{Z_r}{S} = 2\rho c(R + jX)$ is the specific acoustic radiation impedance for radiation to BOTH sides of the membrane $\frac{(N-s)}{m}$;

$$R = 1 - \left[2J_1\left(\frac{2Kb}{\sqrt{\pi}}\right) \Big/ \frac{2Kb}{\sqrt{\pi}}\right] \quad *$$

$$X = K_1\left(\frac{2Kb}{\sqrt{\pi}}\right) \Big/ 2\left(\frac{Kb}{\sqrt{\pi}}\right)^2$$

$$K = 2\pi/\lambda = \omega/c \quad (m^{-1})$$

b is the length of side of square radiator (m)

J_1 and K_1 are Bessel functions of the first order;

$\frac{Z_m'}{S}$ is the entire denominator of equation (4) above $(\frac{N-s}{m})$.

This equation is in the form of $\frac{F/S}{Z_m'/S}$, a force (per unit area) divided by an impedance (per unit area), as was suggested above in 3.10.3.

*The argument of the Bessel functions is ordinarily given as Ka (where a is the radius of a circular element); for the present case of square radiators, we calculate for equivalent area: $b^2 = \pi a^2$ so that

$$Ka \longrightarrow \frac{Kb}{\sqrt{\pi}}$$

3.4.5 The Effective Impedance, Z_m^1

Computations were made for three sizes of square ESUs: Model A is 4 x 4 in; B is 6 x 6 in. and C is 8 x 8 in. All are about 1/2 in. thick and have outside dimensions . 1 in. greater in each case. They weigh 3 - 4 oz. apiece. If we assume a construction, typical of laboratory and production models, in which the membrane is made of 1/2 mil SARAN (specific gravity 1.6) the various components of the impedance may be reckoned for the three sides of ESU at 1000 c/s as follows:

$$\frac{Z_m'}{S} = \left(\frac{j\omega l_m}{S} + \frac{r_m}{S}\;\frac{1}{j\omega C_m S}\right) + 2\rho c(R + jX) - \left(\frac{1}{j\omega}\right)\frac{2E_0^2\epsilon_0}{d^3}$$

$$= j120 + (\text{negligible}) - j48 + \begin{Bmatrix} A:\ 380 + j346 \\ B:\ 692 + j550 \\ C:\ 807 + j408 \end{Bmatrix} \qquad \left(\text{Kg}/\text{m}^2\text{-s}\right) \tag{5}$$

The relative magnitudes of the terms comprising this impedance are such that $Z_m^1 \doteq Z_r$ over most of the relevant frequency range (250 - 7000 c/s). Thus, the velocity of the membrane is determined, not by its own properties, but by its acoustic load. Here arises the first major difference between the ESU and conventional transducers: the impedance of the latter (and hence the diaphragm velocity) is governed, except at the resonant frequency, almost wholly by its own mechanical properties, whereas the ESU is controlled almost entirely by the acoustic loading upon it. The implications of this difference will be made clear later.

3.4.6 Maximum Usable Voltages

It is clear from equ. (4) that for maximum membrane velocity, both the signal and bias voltages should be as large as possible. A practical limit exists, however, on the electric field strength, to avoid electrical breakdown of the air gap plus the usual insulation provisions. This has been found to be about 100 volts/mil ($= 3.94 \times 10^6$ volts/m) for the bias voltage and for the peak signal voltage; this corresponds to 70 v rms/mil($= 2.75 \times 10^6$ volts/m) for the rms signal voltage. These figures will be used for the present calculations. The value of electrode spacing has been chosen as d = 0.017 in. since this, too, is typical of production units. Thus, the maximum bias voltage, applied from membrane to electrode across a distance d, is 1700 volts, dc. The signal voltage on the other hand is applied from electrode to electrode across a distance 2d and may, therefore, have a peak value of 3400 volts or an rms value of 2404 volts.

Notice that the bias and signal voltages enter equ (4) in the form of field strengths (E_0/d and $e/2d$). Thus, although it is possible to use voltages as high as those just mentioned, it is by no means necessary to do so to achieve the same output. If the choice of low frequency limit for the ESU is such that the required sound level can be produced with smaller membrane excursion, then the spacing d can be reduced and the voltages reduced proportionately. This does not decrease the acoustical output (since the

field strength and hence the membrane velocity is unchanged), but the membrane may tend now to strike the stationary electrodes in executing the large amplitudes associated with low frequencies. Of course, it is very seldom that a sound output corresponding to maximum signal and bias voltages is required. This is a matter for design compromise which will be discussed later, in 3. 4 .22.

3. 4 .7 Range of Stable Operation

It is often stated and widely believed that capacitance transducers are inherently non-linear. Condenser microphones are indeed satisfactory only because the diaphragm excursions are small enough to render the distortion products tolerably low. Electrostatic loudspeakers are generally described as having acceptably low distortion (5%) only if the signal voltage is less than 15% of the bias voltage AND if the membrane excursion never exceeds 5% of the membrane-to-electrode spacing. If this were so, there would be no hope of realizing usable sound output with the ESU, at least not in any competitive sense.

But these restrictions apply only to the single-ended type. When advantage is taken of push - pull, constant charge operation, the situation is entirely different. In this case, the bias voltage acts through a very high resistor* to establish on the membrane a constant charge which may be as high as desired, provided the insulation doesn't break down. Then the signal voltage, applied to the two stationary electrodes, establishes a varying field between them, which is uniform throughout the space in which the membrane moves and whose strength is limited again only by the breakdown criterion. There is no reason why the signal voltage may not be several times the bias voltage, since the situation is simply a case of a constant charge moving in a uniform field: either charge or field may be of any desired magnitude. Moreover, those criteria no longer apply which determine for the single-ended ESU the maximum deflection beyond which the membrane collapses onto the stationary electrode (from the attraction of the bias voltage); they are irrelevant here, since if the time constant of the bias voltage circuit is long enough, the electrical forces which would act to cause the collapse do not come into play until the membrane has returned safely out of the danger region. In principle, the membrane excursion may be from electrode to electrode without eit hter danger of permanent collapse or excessive distortion. In practice, of course, it is wise to provide a small margin of clearance to allow for symmetry in production.

3. 4 .8 Maximum Membrane Velocities and Amplitudes

With these design choices made and the values of

Zm^1/S as given in equ. (5) for 1000 c/s :

$$Zm^1/S = \begin{cases} A: & 380 + j\,618 = 726 \angle 58.5^\circ \\ B: & 692 + j\,622 = 930 \angle 42^\circ \\ C: & 887 + j\,480 = 1010 \angle 28.4^\circ \end{cases} \qquad (5')$$

*This resistor is high enough in practice (20-50 megohms) to eliminate any shock hazard from the high bias voltage.

The membrane velocities can be found in each case from equ. (4):

$$v = \left(\frac{2E_0}{Z_m'/S}\right)\left(\frac{\epsilon_0}{d}\right)\left(\frac{e_{sig\,rms}}{2d}\right) = \frac{2(8.85\times10^{-12})(3.94\times10^{6})(2.78\times10^{6})}{Z_m'/S}$$

$$v = \begin{Bmatrix} A: & 0.263 \\ B: & 0.206 \\ C: & 0.189 \end{Bmatrix} \quad (m/s) \qquad (6)$$

Note that the force per unit area and the mechanical impedance are the same in all three cases; the membrane velocity for the larger units is less because the acoustical loading (i.e. radiation impedance) is greater.

These are the maximum permissible rms velocities without risk of insulation breakdown. The corresponding peak amplitudes are given by $X_{pk} = \sqrt{2}\left(\frac{V_{rms}}{j\omega}\right)$ for 1000 c/s :

$$X_{pk} = \begin{Bmatrix} A: & 0.00231'' \\ B: & 0.00179'' \\ C: & 0.00170'' \end{Bmatrix} \qquad (7)$$

Evidently the choice of 0.017 in. for the air gap was extremely generous, provided that no lower frequency than 1000 c/s is required!

3.4.9 The Maximum Pressure at 30 in.

Now that the maximum membrane velocities are known, the maximum axial sound pressure at 30 in. and 1000 c/s can be found from equ. (1):

$$P_{v\,rms} = \frac{\rho f S v_{rms}}{r} = S v_{rms}\frac{(1.2)(1000)}{(0.76)}$$

$$= \begin{Bmatrix} A: & 4.28 \\ B: & 7.6 \\ C: & 12.4 \end{Bmatrix} (N/m^2) = \begin{Bmatrix} 42.8 \\ 76 \\ 124 \end{Bmatrix} (dyne/cm^2) \qquad (8a)$$

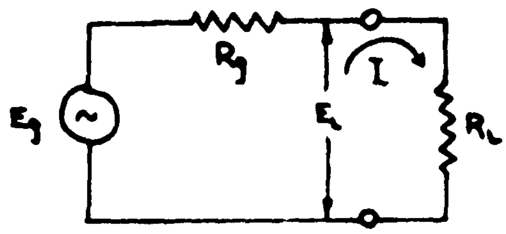
$$= \begin{Bmatrix} A: & 106.8\ db \\ B: & 111.6\ db \\ C: & 116.0\ db \end{Bmatrix} \text{ re } 0.0002\ dyne/cm^2 \qquad (8b)$$

This pressure appears on both sides of the baffle.

Recall that the bias voltage for these pressures is 1700 v DC and the signal voltage 2404 v rms.

3.4.10 Available Power for the ESU

The concept of available power is most clear-cut when both the generator and the load have purely resistive impedance:



The power in the load is $P_L = I^2 R_L = \left(\frac{E_g}{R_g + R_L}\right)^2 R_L$

Maximum power will be available when $R_L = R_g$ and this will be

$$P_{max} = \frac{E_g^2}{4R_L}$$

But when the load is purely reactive, the generator impedance remaining resistive, a certain amount of interpretation is required in defining the equivalent of maximum available power.

In any practical case, the available power from an amplifier is limited by the internal energy dissipation in the output stage.* Thus, it seems reasonable to define available power for the ESU as that power which would be delivered to a matched resistive load when the internal power dissipation in the source is the same as it would be when driving an ESU. Therefore, "matching" for the ESU occurs when the same current is drawn from the source, and this happens when $X_c = \sqrt{3}\, R_g$

Values in each case are as follows:

	I_{rms}	$E_{L\,rms}$	$P_g = I^2 R_g$	$P = E_L I$
MATCHED RESISTIVE LOAD $R_L = R_g$	$\frac{E_g}{2R_g}$	$\frac{1}{2} E_g$	$\frac{E_g^2}{4R_g}$	$\frac{E_g^2}{4R_g}$
ESU LOAD $X_c = \sqrt{3}\, R_g$	$\frac{E_g}{R_g + jX_c} = \frac{E_g}{R_g + j(\sqrt{3} R_g)} = \frac{E_g}{2R_g}$	$\frac{\sqrt{3}}{2} E_g$	$\frac{E_g^2}{4R_g}$	$\frac{\sqrt{3}\, E_g^2}{4R_g}$ (REACTIVE)

*Similar statements hold for other types of source.

Note that for the same power demand from the generator (in terms of internal energy dissipation) the voltage supplied to the ESU is 73% (4.8 db) greater than would be delivered to a resistive load.

The figures of the chart have therefore been calculated as follows: instead of computing the sound output when the same voltage is applied to the ESU as to a conventional speaker, the output was found when the internal power dissipation in the generator is the same. The "matching" relationship, $X_c = \sqrt{3}\, R_g$

is assumed in all computations, so that an amplifier which supplies $E_L = E_g/2$ to a matched resistive load will supply $E_L = \sqrt{3}\, E_g/2$ to the ESU and work no harder, giving the ESU a 4.8 db advantage in signal voltage.

3.4.11 The Axial Pressure at 30 in. for 1 VA Available Power

The electrical impedance for the ESU is approximately that of a pure capacitance. If the capacitance between the membrane and either electrode is C_a, then since two such capacitances appear in series between the two signal electrodes, the impedance is approximately:*

$$X_c = \frac{1}{j2\pi f \frac{C_a}{2}} = \begin{cases} A: & 1.51 \times 10^6 \Omega \\ B: & 0.67 \times 10^6 \Omega \\ C: & 0.377 \times 10^6 \Omega \end{cases} \text{ at } 1000 \text{ c/s} \tag{9}$$

The volt-amperes circulating in the ESU when maximum voltages are applied and when the pressures of equ. (8) are hereby generated are:

$$VA = \frac{e_{su}^2}{X_c} = \begin{cases} A: & 3.84VA = 5.8 \text{ db} \\ B: & 8.65VA = 9.3 \text{ db} \\ C: & 15.4VA = 11.8 \text{ db} \end{cases} \text{ Re } 1VA \tag{10}$$

Then for 1 va circulating in the ESU, the rms axial pressure at 30 in. would be:

$$\underset{\left(\substack{1\text{va in} \\ \text{ESU}}\right)}{P_{ax\ at\ 30''}} = \begin{cases} A: & 106.8 - 5.8 = 101.0 \text{ db} \\ B: & 111.8 - 9.3 = 102.5 \text{ db} \\ C: & 116.0 - 11.8 = 104.2 \text{ db} \end{cases}$$

*This neglects that component of current reflected through the transduction coefficient from the acoustical side of the circuit, but the error is less than 5%.

But for 1 watt dissipated in the generator - - - which is the case when 1 va is supplied to a conventional speaker - - - these values would all be raised 73% or 4.8 db, as discussed in 3.10.10, to give the comparison figures used for the chart:

$$\text{rms } P_{ax} \text{ at } 30'' \left(\begin{array}{c}\text{1 va available}\\ \text{power}\end{array}\right) = \begin{cases} A: & 105.8 \text{ db} \\ B: & 107.1 \text{ db} \\ C: & 109.0 \text{ db} \end{cases} \qquad (11)$$

3.4.12 Peak Instantaneous Pressure at 30 in. for Maximum Rated (Available) Power.

The maximum voltages of 3.10.6 have led to the rms pressures of equ. (8b) with volt-amperes into the ESU given by equ. (10). The peak pressures at 30 in. would be 3 db higher than the res pressures, while the power dissipated in the generator would be 4.8 db less than (i.e. 33% of) that circulating in the ESU:

$$\text{Peak } P_{ax30in.} = \begin{cases} A: & 109.8 \text{ db} \\ B: & 114.6 \text{ db} \\ C: & 119.0 \text{ db} \end{cases} \quad \text{re } 0.0002 \text{ dyne/cm}^2 \qquad (12)$$

$$\text{Max. rated available power} = \left\{\begin{array}{ll} A: & 1.27 \text{ watts} \\ B: & 2.86 \text{ watts} \\ C: & 5.10 \text{ watts} \end{array}\right\} \qquad (13)$$

3. 4 . 13 Retma Rating

The equation for the RETMA rating is [11]

$$G_{SP} = \left(\underset{ax}{SPL} \text{ at } 30'\right) - 10\log\left(\begin{smallmatrix}\text{AVAILABLE}\\ \text{POWER}\end{smallmatrix}\right) - 30 \text{ db.} \qquad (14)$$

For this rating the figures of equ. (11) for the axial sound pressure level at 30 in. corresponding to 1 va of available power were corrected to 30 ft. distance (-21.6 db). With $W_{AS} = 1$, the second term of equ. (14) disappears and the RETMA ratings are:

$$G_{SP} = \begin{Bmatrix} A: & 105.8 - 21.6 - 30 = 54.2 \text{ db} \\ B: & 107.1 - 21.6 - 30 = 55.5 \text{ db} \\ C: & 109.0 - 21.6 - 30 = 57.4 \text{ db} \end{Bmatrix} \qquad (15)$$

If a method is ever found to cancel the static capacitance of the ESU and improve the power factor, G_{SP} will increase enormously. The transduction process itself is more than 99% efficient.

3. 4 . 14 Absolute Efficiency

Because of the highly reactive nature of the ESU (its nearly complete lack of internal electrical resistance means that the real component of its impedance comes from the radiation resistance) it would be grossly unfair to conventional speakers to set up a conversion efficiency comparison based upon the ratio of "energy radiated" to "energy dissipated internally." In the ESU, as Hunt has remarked, "the sound radiation does not need to share the signal energy delivered to the transducer with magnetic-hysteresis losses, eddy-current losses, nor with heat losses in a voice-coil conductor; and since the condenser dialectic is chiefly air, the dielectic losses are - - - significantly lower than for most piesoelectric materials;" the intrinsic conversion efficiency of the ESU is greater than 99%.

However, a more fair comparison can be formulated in terms of an Absolute Efficiency: a ratio of the "radiated energy" to the "total energy dissipated, in both the signal source and the transducer," assuming all units matched to their sources.

The most efficient horn unit listed in the chart is the Altec-Lansing 288B, a 20-lb. unit with a 3 in. voice coil. It produces 119 db at 30 in. on the axis with 1 va input. The directivity index is given as -13 db so the total radiated power may be found as follows:

Avg. SPL = Axial SPL - D.I. = 119 - 13 = 106 db at 30 in., or an average pressure of 4 N/m^2 at a distance of r = 0.76 in; and this, in free space, is an average intensity

$$I_{avg} = \frac{p^2}{\rho c} = \frac{4^2}{414} = 38.5 \text{ mw/m}^2$$

The total radiated power is $4\pi r^2 I_{av} = (12.56)(0.76)^2(0.0385) =$ 0.28 watts for 1 watt input at 1000 c/s. When matched for maximum available power, there will be 1 watt dissipated in the source for every watt delivered to the transducer (which is divided between internal losses in the speaker and radiated sound). Thus, the absolute efficiency is $\frac{0.28}{1+1} = 14\%$, about half the usually quoted figure which does not account for power lost in the source.

For the "C model" ESU the same computation yields:

Axial SPL - D.I. = 109 - 6.1 = 102.9 db at 30 in. for 1 va available power.

This represents an average pressure of 2 N/m^2 (into two hemispheres) and total radiated power of 0.14 watts for 1 va available power. The only dissipated power is the 1 watt lost in the source, so the absolute efficiency is $\frac{0.14}{1.14} = 12.3\%$. As remarked before, if a way can be found to eliminate the purely reactive component of current, the absolute efficiency will rise phenomenally to nearly 100%

3.4.15 Directivity Index

The D.I. is defined as the ratio of the total power actually radiated by a source to that which would be radiated by a point source generating the same pressure at the same point on the axis. The D.I. is always $\leq$ unity; if the speaker is mounted in an infinite baffle and account is taken only of the energy radiated to one side, the D.I. is always ≤ 0.5 ($\leq$ -3 db). The latter case is what is generally computed.

It can be shown that the directivity indes (D.I. of a radiator is directly related to the real part of its acoustical radiation impedances:

$$\text{D.I.} = \frac{\pi}{SK^2} R \tag{16}$$

where S, K and R are as defined in equ. (4).

For the three ESUs the directivity indices calculated for a frequency of 1 KC, are =

$$\text{D.I.} = \begin{Bmatrix} \text{A:} & 0.418 \text{ or } -3.8 \text{ db} \\ \text{B:} & 0.337 \text{ or } -4.7 \text{ db} \\ \text{C:} & 0.246 \text{ or } -6.1 \text{ db} \end{Bmatrix} \tag{17}$$

3.4.16-30 in. SPL Minus D.I.

This quantity is found directly by applying equ. (17) to equ. (11):

$$\begin{aligned} &A.\ 105.8 - 3.8 = 102.0 \text{ db} \\ &B.\ 107.1 - 4.7 = 102.4 \text{ db} \\ &C.\ 109.0 - 6.1 = 102.9 \text{ db} \end{aligned} \qquad (18)$$

3.4.17 Usable Frequency Range

In equ. (5) the relative magnitudes of the various components of Z_m^{1}/S were given for 1000 c/s. It was shown that at this frequency the behavior is dominated by the acoustic impedance term, Z_r/S. This situation prevails for frequencies up to 7000 c/s, at which point the mass reactance of the membrane assumes control and the output begins to drop at the rate of 6 db/oct. If the membrane is made of 1/4 mil. MYLAR (S.G. = 1.38) instead of the 1/2 mil SARAN (S.G. = 1.6), this upper frequency limit is raised from 7000 c/s to 16,000 c/s.

At the low frequency end, the membrane velocity continues to be controlled by the radiation impedance down to frequencies for A, B and C respectively of 240, 190 and 170 c/s. But although the Z_r/S term governs the membrane motion, radiation does not occur efficiently for frequencies for which

$$\frac{Kb}{\sqrt{\pi}} = \frac{2\sqrt{\pi}\, f b}{c} \leq 0.5$$

It is this criterion that was used to set the lower frequency limit of useful operation for each of the ESUs:

$$\underset{\left(\frac{Kb}{\sqrt{\pi}} = 0.5\right)}{f_{min}} = \begin{cases} A: & 476 \text{ c/s} \\ B: & 316 \text{ c/s} \\ C: & 238 \text{ c/s} \end{cases} \qquad (19)$$

3.4.18 Radiation into Small Enclosures

So far, all calculations have assumed the use of loudspeakers radiating into the cockpit area where their sound output competes with a high ambient noise level and must therefore be quite great. An obvious improvement would be to place the transducer inside the pilot's helmet so as to take advantage of the transmission loss of the helmet as it affects the ambient noise level; this would also restrict the space into which the speaker must radiate. A further step in the same direction would be to insert the transducer into the pilot's ear canal. The light weight and simplicity of construction of the ESUs make them very attractive for both these applications, so comparative figures for these conditions were calculated.

3.4.19 Ear Inserts

The design for a typical ear insert device was based on an average ear cavity $V_o = 2$ cm^3 ($= 2 \times 10^{-6}$m^3), a figure derived from physical measurements.[12] The walls of the cavity were considered rigid, as a first approximation. The device was assumed to have a membrane area $S = 0.3$ cm^2 ($= 0.3 \times 10^{-4}$ m^2)

with electrode-to-membrane spacing $d = 0.005$ in. ($= 1.27 \times 10^{-4}$ m.) The effective radius was $r = 3.1 \times 10^{-3}$ m.

If the same limitations on field strength are applied, the membrane velocity can be computed as before from equ. (4), where, for the relevant range of frequencies and ear cavity volumes, $Z_m{}^1$ is contributed almost entirely by the ear cavity stiffness. Radiation to the back side of the diaphragm is either into free space or a large cavity (such as the helmet). When the velocity is known, the pressure in the cavity can be found from equ. (2), which becomes

$$p = v\left(\frac{Z_m'}{S}\right) = \frac{2\varepsilon_0}{\left(\frac{Z_m'}{S}\right)}\left(\frac{E}{d}\right)\left(\frac{e}{2d}\right)\left(\frac{Z_m'}{S}\right)$$
$$= 2\varepsilon_0\left(\frac{E}{d}\right)\left(\frac{e}{2d}\right) \qquad (20)$$

Surprisingly, the pressure in the ear cavity is independent of the impedance of the cavity! This implies, for example, that if the transducer is once calibrated for one size of ear, the same calibration will be valid for ears of other sizes. Furthermore, the pressure is not affected by changes in the density of the medium, and, for constant input voltage, the cavity pressure is independent of frequency, although the current rises as the electrical impedance (capacitive) decreases. These observations depend on the assumption that Z_m'/S is contributed primarily by the stiffness of the cavity. As the frequency increases, however, this impedance decreases and there will be an upper frequency limit above which the mass of the membrane takes over control: above this frequency the pressure drops 6 db/oct. For a 1/2 mil Saran diaphragm and the assumed cavity volume, the critical frequency is 1650 c/s; if the membrane is made of 1/4 mil Mylar, the critical frequency is raised to 2500 c/s.* The lower frequency limit is imposed by the required membrane excursion .

Putting the maximum possible field strengths of 3.10.6 into equ. (20) ($E_0 = 500$ v D.C. ; $e_{sig} = 700$ v rms)

gives a value for ear cavity rms pressure of 191.8 N/m^2 (1918 $dyne/cm^2$) or 139.7 db re 0.0002 $dyne/cm^2$ with 10 mva into the ESU. For 1 milli-voltampere input, the pressure would be 129.7 db and for 1 mw available power, 129.7 4.8 = 134.5 db.

The question should be mentioned of whether or not the use of such high voltages would be hazardous in the case of a transducer to be worn in the ear, considering the danger of permeation with salty perspiration, etc. Of course, they might, but it should be noted that if the signal voltage were decreased 31 db to 20 v rms and the bias voltage reduced 23 db to 35 v DC, the pressure would

*The critical frequency, of course, depends on the size of the ear cavity. A more refined estimate of the effective (rather than the actual physical) volume has been made from impedance measurements at the eardrum, and these indicate an average effective cavity volume of only 0.82 cm^3; this minimum achievable value would raise the critical frequency well above the range that would be used in communications.

still be a respectable 86 db at the ear drum. (The lower frequency limit in this case would be below 10 c/s!) Moreover, since the bias voltage could be (in fact, for the present theory to apply, must be) applied through very large resistor, the hazard from this score would be negligible even if the bias were increased to 100 v, and the SPL to 95 db. The smaller excursions at these lower levels permits reduction of spacing and hence, a still further reduction of voltages.

In any event, the techniques of casting plastic materials are so sophisticated nowadays that a great deal of hazard may be eliminated by skillful design of the transducer and strict production control. The units would be intrinsically cheap enough that considerable care here is still economical. The same considerations apply to the signal source voltage. Even though it should be from a low impedance source the DC resistance can be made very high.

3. 4 . 20 Helmet Insert

This transducer was envisaged as an adaptation of the square model "A" ESU (4 x 4 in.) facing into the MA-1 helmet and covered on the back with a cavity 1 in. deep and lightly filled with Fiberglas.

The relevant volumes were taken as follows:

Volume of MA-1 helmet	7740 cm^3
Volume of padding	2290 cm^3
Volume of average male head	3630 cm^3
Remaining open volume in helmet	1820 cm^3

Thus the ESU faces into a volume $V_1 = 1.82 \times 10^{-3} m^3$ and is backed by a volume $V_2 = 1 \times 4 \times 4$ in. $= 2.61 \times 10^{-4} m^3$. It is again assumed that the cavities act as lumped parameters (in this case pure stiffness) and that thus they present to the ESU an effective volume

$$V_o = \frac{V_1 V_2}{V_1 + V_2} = 2.3 \times 10^{-4} m^3 \text{ which governs the}$$

membrane velocity through the acoustic compliance

$$C_A = \frac{V_o}{\rho c^2} = 1.64 \times 10^{-9}\ m^5/N$$

at 1000 c/s, $Z_m'/S = S/jwC_A = -j\ 10^3$ N-s/m^3, and the corresponding maximum membrane velocity (from equ 4) is 0.192 m/s; the maximum peak displacement is 4.3×10^{-5} m. At 100 c/s the maximum peak displacement is 4.4×10^{-4} m; that is, with the 0.017 in. ($= 4.32 \times 10^{-4}$m) spacing between membrane and electrodes, it just "bottoms." This velocity, however, yields a pressure within the helmet of

$$p = \frac{vS}{j\omega C_A'} = 2\varepsilon_o \left(\frac{E_o}{d}\right)\left(\frac{e}{2d}\right)\left(\frac{C_A}{C_A'}\right) \qquad (21)$$

where C'_A is the acoustical compliance of the helmet cavity only

(= 7.94 x 10^{-3} $\frac{M^2 s}{Kg}$ at 1000 c/s). At 1000 c/s this rms pressure is 24.2 N/m^2 or 121.6 db, and again, the pressure in the cavity is independent of frequency and of the properties (within a certain range) of the medium.

The electrical power dissipated within the source to achieve this SPL is 2.41 watts. Assuming that the SPL inside the helmet never need be more than 100 db, the bias and signal voltages can be reduced 22 db between them - - - e.g. -10 db for E_o and -12 db for e_s - - - to give E_o = 535 v DC and e_s = 600 v rms. The maximum peak displacement at 100 c/s for these decreased voltages is reduced to 0.35 x 10^{-4}cm. and therefore the electrode spacing can be reduced to about 0.43 x 10^{-4} cm. As a result of this, both E_o and e_s can be further proportionately reduced another 20 db apiece to E_o = 54 v DC and e_s = 60 v rms to give a SPL of 100 db from the single 4 x 4 in. transducer.

The original(maximum) voltages yielded a SPL = 121.6 db for 2.41 watts available, or 117.6 db for 1 watt available. At this 1 watt level and with the reduced spacing above, the low frequency limit would be 50 c/s.

A useful figure for comparison in both the cases of the ear and helmet inserts is the SPLproduced by an input of 1 volt at an 8-ohm impedance level. An ideal transducer is assumed to accomplish the stepup of impedance to match each unit:

$$\left(\frac{N_1}{N_2}\right)^2 = \left(\frac{E_1}{E_2}\right)^2 = \frac{Z_1}{Z_2}$$, so that the secondary

voltage in each case is $E_2 = \sqrt{Z_2/Z_1}\, E_1$ for 1vrms applied to the primary=

EAR INSERT

$Z_2 = 5 \times 10^7 \Omega$

$Z_1 = 8 \Omega$

$E_2 = \sqrt{\frac{5 \times 10^7}{8}} \times 1 = 2500$ v. rms

(N. B., this voltage exceeds the breakdown limit).

$p = \frac{2500}{1400} \times 3835 = 6850$ dyne/cm^2

= 151 db

HELMET INSERT

$Z_2 = 1.36 \times 10^6 \Omega$

$Z_1 = 8 \Omega$

$E_2 = \sqrt{\frac{1.36 \times 10^6}{8}} \times 1 = 412$ v. rms

$p = \frac{412}{4760} \times 483 = 41.7$ dyne/cm^2

= 106 db

3.4.21 Variation of Radiated Pressure with Altitude

Mention should be made of an unusual corollary of the fact that the acoustical loadings govern the membrane velocity of the ESU. It has already been seen in the case of transducers operating into closed cavities that the resulting pressures are independent of the cavity size and the properties of the medium within reasonable variations.

It can further be shown for the elements radiating into a field that, whereas the acoustical output of virtually every other source of sound diminishes as a result of changes in the properties of the medium as the altitude is increased, the ESU output increases.

For example, let it be assumed that the average sound outputs of several different types of sound source are equal at sea level. Then, if with the same signal voltage applied, they are all taken to 35,000 ft. altitude, the sound outputs would compare with the common sea level output as follows:

Far Field	ESU speaker baffled	6 db
	ESU speaker unbaffled	8.5
	Dynamic speaker baffled	-6
	" " unbaffled	-3.5
	ESU (near field)	0
	Dynamic speaker (near field)	-12
	Aerodynamic noise	-8
	Boundary layer noise	-12

In the case of decompression at high altitude, the ESU transducer is the only type of unit for use in cockpit communications for which the signal-to-noise ratio is not seriously impaired - - - in fact, with the ESU it would be greatly improved, which is a highly desirable feature in case the cause of the decompression is a sizable hole which itself will be an intense noise source.

3.4.22 Reduction of Operating Voltages

The voltages assumed throughout the calculations have been quite high as a result of using the maximum practical field strength and selecting an electrode spacing typical of a commercially produced unit which has been successfully used in all climatic ranges of temperature and humidity and at altitudes up to 10,000 ft.

It was shown, in the case of the ear and helmet inserts, that wherever requirements for low frequencies and/or high SPL, are not severe, the electrode spacing and hence the voltages may be reduced while maintaining the same electric fields in the A, B and C. models, as well.

For example, instead of choosing the 0.017 in. spacing as a standard for all three models, it would be better to design in terms of the peak excursion associated with the maximum pressures of equ. (8) at the low frequency limits of equ. (19):

	A	B	C
$p_{30''}$ =	107 db.	112 db.	116 db.
V_{rms} =	0.263 m/s	0.206 m/s	0.189 m/s
f_{c_0} =	476 c/s	[illegible]16 c/s	238 c/s
FOR SPACING DESIGN USE f_0 =	450 c/s	300 c/s	200 c/s
THEN AT f_0: X_{pk} =	1.31×10^{-4} m	1.53×10^{-4} m	2.1×10^{-4} m
CHOOSE SPACING d =	1.5×10^{-4} m	2.0×10^{-4} m	3.0×10^{-4} m
WHICH LEADS TO E_0 =	590 v D.C.	786 v D.C.	1180 v D.C.
e_{sig}	835 v rms	1113 v rms	1670 v rms

If lower SPLs are required, the voltages may be doubly reduced: in the first place, to reduce the field strengths required and in the second, because with reduced membrane excursion the spacing can be decreased and the voltages further reduced to maintain the same field strengths.

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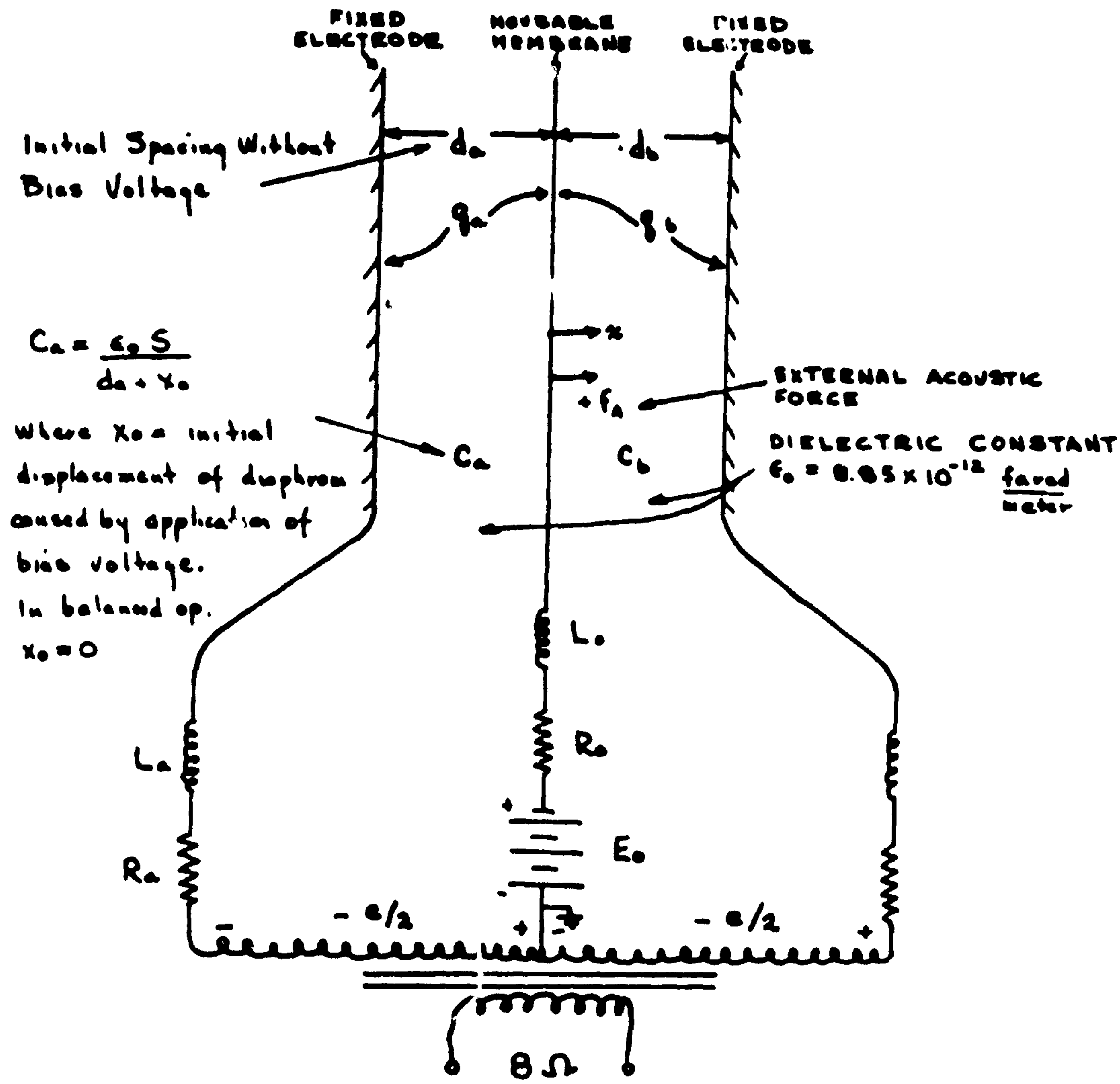
Figure A3-1



In Balanced Operation:

$d_a = d_b = d$

$q_a = q_b = q$

$C_a = C_b = C_0$

$L_0 = L_a = L_b = 0$

$R_a = R_b = 0$

R_0 very large

Chart A3-1

ESU Model	Size	Capacitance mmf	Retma Pressure* Rating (DB)	Axial SPL at 30" for 1 watt available power	Peak inst. SPL at ** 30" for max. rated available power	Directivity Index** (db)	30" SPL - DI (db)	Freq. range limits (cps)
A	4"x4"	211	54.2	105.8	109.8 @1.27 w	-3.8	102.0	476
B	6"x6"	475	55.5	107.1	114.6 @2.86 w	-4.7	102.4	316
C	8"x8"	845	57.4	109.0	119.0 @5.1w	-6.1	102.9	238
Ear Insert	0.3 cm^2	211	-	[illegible]4.5 (1mw)	-	-	-	<100
Helmet Insert	4"x4"	211	-	117.6	-	-	-	<100

APPENDIX 4.0

STUDY OF MICROPHONE SYSTEMS

BY

WESTERN ELECTRO-ACOUSTIC LABORATORY

APPENDIX 4

STUDY OF MICROPHONE SYSTEMS

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OUTLINE OF FIGURES IN APPENDIX 4

Tables

Appendix

4.1 Evaluation of Microphone Systems on the Basis of Physical Measurements

4.1.1 Introduction

In studying various methods of speech pickup it is important that the evaluation technique be diagnostic, i.e. it should explain the reasons for the success or failure of a given system as well as simply give an overall evaluation of relative performance. If possible the evaluation technique should give a prediction of the intelligibility of speech in noise. The diagnostic criteria should minimize uncertainties associated with individual differences, as in articulation testing.

Physical measurements have, therefore, been made to determine the acoustic properties of each system. Examples of such properties are real voice response, noise-cancelling properties, signal-to-noise ratio, etc. An examination of each of these properties is necessary to evaluate each system individually and all systems may be compared on the basis of their physical properties.

4.1.2 Rating Scheme

4.1.2.1 Purpose

Since a large number of microphone systems were tested, a means of rating the performance of each was essential. A technique for rank ordering the systems relative to the performance of a reference system was established.

4.1.2.2 Criteria

It has been shown that the signal-to-noise ratio by octave bands at the microphone may be used to predict a system's effectiveness in noise. A by-product of the physical measurements undertaken was a measurement of signal-to-noise ratio for a given speaking level in the standard 120 db jet noise field. Hence, the signal-to-noise ratio by octave bands was chosen as a convenient means of comparing systems.

4.1.2.3 Reference System

We found it convenient to rate a microphone system's performance in relation to a reference system. The reference system chosen was a Western Electric 640AA with a probe placed touching the speaker's lips. This choice was not arbitrary since this reference system has several advantages: (1) it may be calibrated absolutely; (2) its performance is easily comparable to other systems; (3) it is highly stable in characteristics; (4) since the probe opening is very small, its position can be accurately defined.

4.1.2.4 Technique for Rating Microphone Systems

As stated in paragraph 4.1.2.2, the criteria chosen are signal-to-noise ratio by octave bands. If now the signal-to-noise ratio of a system is compared to the signal-to-noise ratio of the reference system, we may obtain the "improvement in signal-to-noise ratio of the system relative to the reference system." This is valid only if the same speaking effort and

noise level are used for both systems. The procedure for determining this improvement in signal-to-noise ratio is shown graphically in Figure A4-1.

4.1.3 Results of evaluation using the rating scheme

As described above, the end result of the physical measurements is the improvement in signal-to-noise ratio by octave bands of the system under consideration relative to the reference system. It has been shown that signal-to-noise ratio measurements taken in the 300-600, 600-1200, 1200-2400, 2400-4800, and 4800-9600 cps octave bands (Ref. 1) give the best reliability in predicting the intelligibility of communication systems. The 1200-2400 and 2400-4800 cps bands are weighted twice in this scheme.

Let

S/N in	300-600	cps	band	equal	A
	600-1200	"	"	"	B
	1200-2400	"	"	"	C
	2400-4800	"	"	"	D
	4800-9600	"	"	"	E

$$\frac{A - B - 2C - 2D - E}{7} = R$$

We will call the result R a figure of merit for the microphone systems tested based on physical measurements. In our measurements the quantities A to E are actually improvement in signal-to-noise ratios relative to the reference system; hence, R is also relative to the reference rating.

In Table I the improvement in signal-to-noise ratios obtained from measurements outlined in Appendices 4.2 - 4.7 is averaged for two subjects, TW and MG.

It should be stressed that these results take account only of a known major factor in performance and may not correctly rank order a system's effectiveness in noise because no account is taken in these physical measurements of speech quality. In other words, a system which distorts speech may actually give poorer performance than a system with lower improvement in signal-to-noise ratio but no speech distortion.

Appendix

4.2 Physical Studies of Pressure Microphone

4.2.1 Introduction

A pressure microphone is one that responds to changes in sound pressure. A pressure microphone was constructed using a Western Electric 640AA microphone with attached probe tube. Measurements were made with this microphone close to the lips with no external shield over the subject's face, in a noise shield covering his mouth and nose, and in a helmet which completely covered his head.

This close-talking exposed pressure microphone is the reference system described in section 4.1.2.3 of the Appendix. The performance of other systems will be compared to it.

4.2.2 Description of Probe Pressure Microphone

Two probe tube microphones were built for use in our investigation of microphone placement for pickup of speech. These two probe microphones were intended for use both as close-talking pressure microphones, and also as components of an experimental gradient microphone.

Several microphones were considered as the basic pressure transducer for these probe microphones, and the WE 640AA was chosen because of its availability at this laboratory and also because of certain physical features which facilitate the attachment of the probe. The threads which normally hold the gridded cap on the 640AA may also be used to retain an adaptor plug to which the probe tube is attached.

The probe tubes themselves are three-inch lengths of brass tubing .125 inch O. D. and .090 I. D. These tubes are soldered to brass adaptor plugs which fit in front of the 640AA microphone. Figure A4-2 shows the disassembled parts of the probe microphone system and Figure A4-3 shows the system as used on a subject.

Several response curves were measured on the probe microphone to determine the optimum amount of damping necessary to smooth the response without excessively reducing the sensitivity of the probe at high frequencies. Figure A4-4 illustrates the effect of damping on the microphone response.

To achieve the "flat" frequency response ultimately desired, a pair of equalizers were built into aluminum cans, fitted with the proper plugs so that they may be attached to the microphone cable socket on a Western Electro-Acoustic Laboratory Type 100D Condenser Microphone Complement, necessitating no internal wiring changes in this piece of equipment. Figure A4-5 is the insertion loss of the equalizer. In Figure A4-6 the response characteristic of the equalized microphone with optimum damping is shown.

4.2.3 Pressure Microphone in the Open at the Lips

4.2.3.1 Maximum vocal effort

If a subject is placed in an extremely noisy environment, he will raise his voice, if possible, until he can be understood. There is, of course, a limit to the amount of speech he can produce. No data was available on the maximum sound pressure level which a speaker can maintain for a short duration as measured by a microphone very close to the lips. Accordingly, this data was taken on eight personnel in our laboratory, all male, between the ages of 20 and 41. The results are shown in Figure A4-7 . A WE 640AA microphone was placed touching the mouth, slightly above the lips to be free from breath blast effects. The "Joe . . . lawn" sentence was repeated and an average reading was taken over the two sentences. The Sound Pressure Levels given represent the rms pressure averaged over two "Joe . . . lawn" sentences.

4.2.3.2 Speech spectrum vs effort

In Figure A4-8 speech spectrum for various speech efforts ranging from quiet speech to maximum effort is shown for one subject, TW. These measurements were made with the pressure probe microphone described in section 4.2.2 above. These measurements allow one to determine the signal-to-noise ratio for any noise field as a function of speaking effort, by octave bands.

4.2.3.3 Soundpressure level vs subjective speech effort

Description of test:

A subject was seated in a sound proof room with highly absorptive walls with his lips one foot from a WE 640AA microphone directly in front of him. The individual participating was asked to speak at the following seven different vocal efforts:

1. whisper
2. very soft
3. lowered
4. normal conversation
5. raised
6. very loud
7. maximum effort

These efforts were called for in random order and the effort exerted was completely subjective and left up to the person speaking. The sentence "Joe . . . lawn" was repeated several times at each effort and the output of the microphone was read on a Western Electro-Acoustic Laboratory Sound Analyzer with the meter movement damped to read the long time average level for the sentence. The talker could not see the meter.

Results

The results of these tests are shown in Figure A4-9. Ten male speakers were used in these tests, each one tested approximately three times at each level. The spread in results at a given effort for a particular individual was surprisingly small.

4.2.3.4 Response of pressure microphone as a function of distance from lips.

In order to obtain as high a signal-to-noise ratio as possible, a pressure microphone should be placed as close to the source - in this case, the lips - as possible. A study in the decrease in speech sound pressure level as a function of distance from the lips for a constant speaking level was made. Results of this study are shown in Figure A4-10.

The position at the lips denotes the closest position the microphone could be to the lips and still not seriously impair speech. The lips were actually contacting the microphone. A separate fixed microphone enabled the speaker to maintain a constant speech level as the travelling microphone was moved.

The results in Figure A4-11 will be compared with other moveable microphones discussed in later sections.

4.2.4 Pressure Microphone in Noise Shield

4.2.4.1 Description of Noise Shield

A noise shield was made of a hard Fiberglas shell with a thickness ranging from 3/16" to 5/16" depending on position. The shell encloses the speaker's nose so as not to lose nasal sounds. Breath opening can be provided in several manners: (1) A narrow slot between the lower lip and chin which thus leaves the chin free of movement; or (2) a 3" long 3/8" I. D. tube protruding from the bottom of the shield. The interior of the shield is lined with 1/4" ultrafine Fiberglas aircraft blanket. The volume of the mask is 310 cc. When it is placed on a subject, the unoccupied volume is approximately 180 cc.

Two small holes on either side of the shield allow entry of the probe microphones. These holes have been fitted with soft rubber grommets for vibration isolation. Two 3/16" thick Plexiglass windows were made to allow viewing inside the noise shield when it is on an observer. A photograph of the probe microphone in the noise shield is shown in Figure A4-12.

4.2.4.2 Apparatus

The pressure microphone used consisted of the WE 640AA with probe tube attached described in section 4.2.2.

Some modification of the probe assembly was necessary when the microphone was used inside the noise shield. A possibility of a flanking acoustical and/or mechanical path

exists when the probe is placed in a noise shield since part of the microphone system (i.e., part of the probe, 640AA and preamplifier shell) is directly exposed to external noise, while the probe opening may be in a noise field which is 30 to 40 db lower.

After much experimentation the apparatus of Figure A4-13 was chosen. The following precautions were observed:

(1) The joint between the microphone and probe tube was covered with a thick grease to prevent the entry of acoustical signals through the leak.

(2) The preamplifier was surrounded with an additional shell filled with Fiberglas to prevent tube microphonics due to external noise.

(3) The microphone assembly was hung by Neoprene "O" belts from the ceiling for vibration isolation.

(4) Two soft rubber grommets through which the probe tubes pass were placed on the noise shields. This aids in isolating the probes from the shield, since the fit is quite loose. This leak is then sealed with lubricant.

The attenuation of the noise shield under several conditions is shown in Figure A4-14. Note that the measurement on the dummy head indicates that the shield is potentially capable of approximately 40 db of attenuation.

The decrease in attenuation on human subjects may be attributed to the passage of sound through the unprotected fleshy portions of the face and neck into the shield, and of course the effect of any air leak to the outside.

4.2.4.3 Long time average speech spectrum in noise shield

Long time average speech spectrum using the 640AA pressure probe microphone in the noise shield was measured for several speakers. A typical result is shown for talker TW in Figure A4-15. A level approximately 6 db over normal conversation level is used. For comparison, the speech spectrum in the open is also shown for the same speaking effort, using the reference system. Note the increase of speech sound pressure level in the shield at the low and middle frequencies. We refer to this increase in level as speech amplification.

4.2.4.4 Improvement in signal-to-noise ratio of a pressure microphone in the noise shield relative to the reference system

The data outlined in sections 4.2.4.3 and 4.2.4.4 may be combined to obtain the improvement in signal-to-noise ratio of the pressure microphone in the shield relative to the reference system. This improvement in signal-to-noise ratio is the summation of the attenuation of the noise shield and the speech amplification. The results for one subject, TW, are shown in Figure A4-16.

4.2.5 Pressure Microphone in MA-1 Helmet

4.2.5.1 Description of helmet

The MA-1 helmet is used in conjunction with a full pressure system for high altitude flying. It is roughly spherical in shape with a canvas and rubber collar which attach to the pressure suit. It contains a face plate which may be opened or closed. Photographs of the helmet and collar are shown in Figures A4-17 and A4-18.

To facilitate measurements, two small holes were drilled on the front of the helmet just below the face plate to allow entry of the 640AA probe microphone. Clay and lubricant were then placed around the probes at the point of entry to seal the leak. A photograph of the helmet worn on a human subject showing the microphone may be found in Figure A4-19.

4.2.5.2 Attenuation of the MA-1 helmet

The attenuation of the MA-1 helmet was measured as the difference in sound pressure level at the lips in the open and in the helmet when the subject was placed in a jet noise field.

The effect of the material at the base of the helmet which fits around the wearer's neck is shown in Figure A4-20. Curve 4 was obtained by removing the material and resealing the helmet to a heavy steel plate with clay. It is easily seen that the material used to link the helmet to the pressure suit poses a serious limitation on the attenuation attainable in the helmet.

The effect of opening the exhaust valve is shown in Figure A4-21. This is not now a limitation, since the collar now imposes the limit to attenuation; but if the attenuation through the collarwere increased, this leak must be corrected. A simple acoustic filter could be incorporated to insure adequate attenuation and ventilation when the valve is open.

The attenuation of a later model of the MA-1, the MA-3 helmet, is shown in Figure A4-22.

4.2.5.3 Long time average speech spectra in helmet

The sentence "Joe . . . lawn" was repeated several times and recorded. One sentence was made into a tape loop and analyzed by octave band, using a full wave averaging voltmeter with a very long time constant. Using this technique, the long time average (LTA) speech spectrum was obtained. The speaking level was approximately 6 db above normal conversational level.

Results are shown in Figure A4-23 for one subject, TW. Also plotted for comparison is the LTA speech spectrum obtained with the reference system for the same speaking level. Note the slight increase in level in the low and middle frequency bands. This speech amplification is compared with the speech amplification in the noise shield (section 4.2.4.4) in Figure A4-24.

4.2.5.4 Improvement in signal-to-noise ratio of the pressure microphone in the MA-1 helmet relative to the reference system

Combining results from sections 4.2.5.2 and 4.2.5.3, we may obtain the improvement in signal-to-noise ratio of the pressure microphone in the MA-1 helmet over the reference system. This is shown in Figure A4-25.

Appendix

4.3 Physical Studies with Gradient Microphone

4.3.1 Introduction

A pressure-gradient microphone is one that responds to a difference in pressure at two closely-spaced points. This microphone discriminates against noise coming from a distant source with respect to sound coming from a close source. A laboratory gradient microphone was developed using two pressure probe microphones. Measurements were made to evaluate the performance of this microphone with respect to its noise cancelling properties and its effect on speech in the open, in a noise shield, and in a helmet.

4.3.2 Description of Gradient Microphone and Associated Instrumentation

The gradient microphone was constructed using two WE640AA probe microphones (see section 4.2.2) mounted with only a small distance between the probe tube openings. This distance may be easily changed. The pressure difference was measured by subtracting the outputs of the two probe microphones electrically with a transformer. The electrical block diagram is shown in Figure A4-26. A photograph of the microphone is shown in Figure A4-27.

4.3.3 Gradient Microphone in the Open at the Lips

4.3.3.1 Noise cancellation

The construction of the gradient microphone described above allows us to measure the noise cancellation of the microphone directly. Let the difference in pressure of the two microphones equal δp. Let the pressure of one microphone equal p. The difference p - δp, expressed in db, gives the noise cancellation of the gradient microphone.

Measurements were made to determine the noise cancellation as a function of probe spacing. Results are shown in Figure A4-28 for spacings of 3/16", 5/16", and 1/2". The effect of a baffle between the two probes is shown in Figure A4-29.

4.3.3.2 Long time average speech spectra

Long time average speech spectra for probe spacings of 3/16", 3/8", and 3/4" are shown in Figure A4-30. Also plotted on this graph for comparison is the LTA speech spectrum for the reference system. All spectra are at the same speaking level, approximately 6 db above normal conversational level.

4.3.3.3 Gradient speech output as a function of distance from the lips

A constant speech level was maintained as the probe gradient microphone was moved from the lips. LTA speech spectra are shown in Figure A4-31 for distances up to 1"

away from the lips. A constant probe spacing of 1/4" was maintained.

4.3.3.4 Improvement in signal-to-noise ratio relative to reference system

By combining the data from sections 4.3.3.1 and 4.3.3.2 we may obtain the improvement in signal-to-noise ratio over the reference system for probe spacings of 3/16", 3/8", and 3/4". These are shown in Figure A4-32. These results show that the improvement in signal-to-noise ratio improves at middle and high frequencies as the probe spacing decreases.

The noise cancellation of a gradient microphone remains constant as it moves from the lips to a position 1" away from the lips in our noise enclosure. However, as seen in Figure A4-31, the response of the gradient microphone decreases relative to the reference system as the probe pair are moved away from the lips. Drawing upon results from Figures A4-20, A4-28 and A4-31, the improvement in signal-to-noise ratio of the gradient microphone relative to the reference system as both are moved from the lips may be calculated. This is shown in Figure A4-33. These results show that by the time the gradient microphone is moved only 1/2" from the lips, the gradient microphone shows little improvement over the pressure microphone in the 1200-2400 cps band and higher. Moving it out to 1" shows no improvement in the 600-1200 cps band either.

We may therefore conclude that if maximum benefit is to be derived using a gradient microphone, it must be used as close to the lips as possible, and if the distance is as great as 1", it has little virtue.

Further degradation in performance occurs when the gradient microphone is placed above or below the lips or if the angle between the axis of the gradient microphone and the direction of incident sound is other than 90°. These positions were not analyzed in detail, but showed qualitatively great loss in performance.

4.3.4 Gradient Microphone in Fiberglas Noise Shield

A photograph of the gradient microphone in the noise shield is shown in Figure A4-34.

4.3.4.1 Noise Cancellation

Measurements were made to determine whether the noise-cancelling properties of a gradient microphone are effective when placed in a noise shield. The noise shield described in Appendix section 4.2.4.1 was used. Typical results are shown in Figure A4-35. These results indicate that a gradient does develop under these conditions in the noise shield which decreases the noise cancellation at high frequencies.

4.3.4.2 Long time average speech spectra

Long time average speech spectra for a probe spacing of 1/4" were measured in the noise shield. A typical result is shown in Figure A4-36. Also plotted on this graph for comparison purposes are LTA spectra for the reference system and the pressure microphone. These are all at the same speaking effort. These results indicate that the gradient microphone decreases some of the speech amplification occurring in the noise shield.

It was impossible to perform a controlled experiment to determine the effect of moving the gradient microphone from the lips as the subject spoke at a constant level. However, it was noted that the positioning of the microphone relative to the lips was extremely critical; at least as critical as in the open (see Appendix section 4.3.3.3). Therefore, for optimum performance, the gradient microphone should be as close to the lips as possible when used in a noise shield.

4.3.4.3 Improvement in signal-to-noise ratio relative to the reference system

By combining the data from sections 4.2.4.1, 4.3.4.1 and 4.3.4.2, the improvement in signal-to-noise ratio of the gradient microphone in the noise shield relative to the reference system is obtained. This is shown in Figure A4-37 for subject TW. This is typical of results with other talkers.

4.3.5 Gradient Microphone in MA-1 Helmet

A photograph of the gradient microphone in the MA-1 helmet is shown in Figure A4-38.

4.3.5.1 Noise Cancellation

Measurements were made to determine whether the noise cancelling properties of a gradient microphone are effective when placed in a helmet. A typical result is shown in Figure A4-39. This indicates that the noise cancelling property is approximately the same in the helmet as it is in a diffuse field.

4.3.5.2 Long time average speech spectra

Long time average speech spectra were measured in the MA-1 helmet and are shown in Figure A4-40. A probe spacing of 1/4" was used. Also plotted on this graph are the reference system and a pressure microphone for the same speaking level.

4.3.5.3 Improvement in signal-to-noise ratio relative to the reference system

Combining data from section 4.2.5.1, 4.3.5.1, and 4.3.5.2, the improvement in signal-to-noise ratio of the gradient microphone in the MA-1 helmet relative to the reference system is obtained. A typical result is shown for subject TW in Figure A4-41.

Appendix
4.4 Speech Pickup in the Ear

4.4.1 Introduction

Investigators have reported picking up intelligible speech in the ear.[3] In this section we report certain physical measurements to determine the absolute sound pressure level in the ear. These measurements are combined with data to determine how the ear microphone ranks with other systems on the basis of signal-to-noise ratio.

4.4.2 Apparatus

4.4.2.1 Insert Device

A Harvintip was chosen to couple the microphone to the ear. It was chosen because it is a generalized insert device, its characteristics are well known, and it was readily available.

4.4.2.2 Microphone

It was considered essential that the performance of this system be comparable on an absolute basis with all other systems. The choice of a microphone was dictated by the requirements that it be easily calibrated, have good frequency response, and relatively small size. The 640AA microphone met the first two requirements, and through the use of an auxiliary headband to support the microphone, the size problem was overcome. A special adapter replaces the grid of the microphone and attaches to the Harvintip. The apparatus is shown in Figure A4-42. A calibration of the 640AA on the Harvintip in a coupler simulating the ear canal is shown in Figure A4-43. The apparatus for making this measurement is shown in Figure A4-44. A photograph of the assembly on a subject is shown in Figure A4-45.

4.4.3 Airborne Speech in Ear Canal

Speech in the ear canal may arrive through two general paths, i.e. path P_1 external airborne speech from lips entering canal and second path P_2 solid borne speech through head entering the canal. In this section we determine the level in the ear canal due to solid borne and airborne speech only. Since we are here only interested in the solid borne speech, we must determine the contribution due to airborne speech in the ear canal.

It is helpful to define the following quantities which apply to Figure A4-46.

Let L = speech level at the lips in db
O = speech level outside the ear canal due to airborne speech in db
I = speech level inside ear canal due to airborne speech plus bone conducted speech in db
A = speech level inside ear canal due to airborne speech only
B = attenuation of Harvintip assembly to airborne noise

First, let us consider the level of airborne speech alone in the ear canal for a given speaking level. For a given level L, there will be a level O which is lower than L due to the path length P_1 and any directivity at the mouth, shown in Figure A4-47. The level inside the ear canal is also diminished by quantity B, the attenuation of the Harvintip assembly. (Figure A4-48) The level O-B=A, the level of airborne speech in the ear canal. The technique for measuring B is shown in Figure A4-46, Part I. Now with the same speaking level L we measure the speech level I inside the ear canal. (Figure A4-46) Now if $I \gg A$, it is clear that the solid borne speech is greater than the airborne speech. The contribution of the solid borne speech, of course, depends upon the quantity I - A in db. If I - A = 16 db, then airborne speech is contributing only about .1 db, whereas if I - A = 6 db, then the airborne speech is contributing 2 db. Measurements were undertaken to determine the quantity I - A for each speaker used. This quantity may be thought of as a solid borne/airborne speech ratio. A typical result is shown in Figure A4-49.

4.4.4 Long Time Average Speech Spectrum in the Ear

Immediately after the measurements to determine the level of airborne speech in the ear were made, speech at the lips and in the ear were recorded. Although both recordings were not made simultaneously, the same speaking level was maintained through the use of a monitor microphone 1' away. Results are averaged in Figure A4-50 for three speakers. Using the technique described in section 4.4.3, it was possible to say that the airborne speech was not contributing significantly to the level in the ear.

4.4.5 Difference Between Level at the Lips and the Level in the Ear for Same Speaking Effort

This difference, taken from data not reported here, is averaged for the speakers and is shown in Figure A4-51. We find that the speech level in the ear is lower than that at the lips, especially at high frequencies. Since this is in fact a loss of signal strength, external noise must be attenuated by the same amount if an ear pickup is to have the same S/N ratio as an open pressure microphone at the lips. The difference curve of Figure A4-51 also indicates what the complementary microphone response of an ear transducer should be to match the characteristic spectrum at the lips. This is shown in Figure A4-52.

4.4.6 Noise Exclusion with Various Ear Protectors

As was pointed out in section 4.4.5, since the speech level is low in the ear, considerable noise exclusion must be employed to achieve a satisfactory S/N ratio. Several alternatives are possible:

1. An insert tip, such as Harvintip
2. Aninsert tip plus ear muffs
3. An insert tip plus helmet

Representative data on the noise exclusion of these devices is shown in Figure A4-53.

4.4.7 Improvement in S/N Ratio

Combining the speech data from sections 4.4.4 and 4.4.5 and with the noise exclusion data of various ear protectors of section 4.4.6, we may compute the improvement in S/N ratio of an ear microphone over the reference system, i.e. a pressure microphone at lips. Results are shown in Figure A4-54. The results are averaged for three observers.

This improvement in signal-to-noise ratio may be somewhat misleading since it does not take into account the quality of speech in the ear. The relative merit of the ear pickup will be ascertained only with articulation testing which is discussed in section 4.8.

4.4.8 Ear Microphone Used in Articulation Tests

4.4.8.1 Response

The bulky size of the 640AA on the Harvintip made it unfeasible to use the device under the ear muff. However, we wished to test the ear microphone under a muff as part of our articulation testing program. We, therefore, sought another transducer other than the 640AA, since the 640AA had served its function, i.e. an absolute calibration of the sound pressure level in the ear.

A small American dynamic microphone pressure unit was available in this laboratory which could be attached to a Harvintip. It is shown in Figures A4-42 and A4-55. Its relative calibration is shown in Figure A4-56. The decrease in sensitivity at low frequencies corresponded nearly to the equalization required at low frequencies (Figure A4-52). Some additional equalization was needed above 2000 cycles, however. (Figure A4-52)

Long time average speech spectra using this equalized dynamic microphone on subject MG are shown in Figure A4-57. It compares favorably to the spectrum shape at the lips. Similar agreement occured on other subjects.

4.4.8.2 Improvement in signal-to-noise ratio of dynamic ear pickup over pressure microphone

The improvement in signal-to-noise ratio of the dynamic ear microphone using just the Harvintip and Harvintip plus Clark Muff Model #372-8A-F (Figure A4-58) over an open pressure is shown in Figure A4-59. These were computed by comparing the S/N of the ear pickup to the S/N of the open pressure microphone in the same noise field and with the same speaking level.

Appendix

4.5 Forehead Pickup

4.5.1 Introduction

Speech pickup at many anatomical locations has been described in the literature. Since our time was limited, we chose to make a quick survey of these locations, and if any looked promising, to explore this location more fully. A simple coupler was made using a 640AA microphone which could be placed on the body. On the basis of quality the forehead appeared the best.

4.5.2 Measurement of skull vibrations with a cavity-coupled condenser microphone, with special attention to bone-conducted speech

For many years throat microphones have been used for picking up speech. The quality is poor and it is known that a suitable transducer, placed on the forehead, will give a higher intelligibility. Design of such a transducer requires knowledge primarily of the spectrum of the skull vibrations, and it is shown in the following how a Western Electric 640AA condenser microphone can be used for absolute measurement of the vibration spectrum.

4.5.2.1 The principle in the measurements is simply to place a rigidly held coupler on the forehead in such a manner that the forehead constitutes one end of the coupler, while the 640AA microphone is placed in the other end, (see Figure A4-60). Skull vibrations will produce pressure variations in the cavity, which are picked up by the microphone.

Although a simple device, several factors must be taken into account when such a coupler is designed. They are all concerned with the compliance of the skin.

The problems are most easily interpreted by a discussion of Figure A4-61, where two basic coupler designs are shown. In both designs the cavity is made up by the coupler (M_R) and a diaphragm (C_D, M_D). The microphone (included in M_R) is omitted in the drawings for the sake of simplicity. In the left hand figure the coupler contacts the skin (C_S M_s) by the diaphragm only, while in the right hand figure it contacts the skin partly by the edges, partly by the diaphragm.

In each drawing three forces are acting upon the cavity, namely

f = the force due to internal vibrations (in this case, the speech generation in the throat and in the mouth cavity)

f_R = the force on the coupler unit due to an external sound field (f_R) = p x f (surface), where p is the sound pressure and f (surface) is proportional to the equivalent surface area, upon which p is acting.

f_a = the force on the diaphragm due to sound vibrations conducted through the skin

We want the largest ratio between f and $f_R + f_s$. An analysis of the design problems is most easily carried out by first studying the equivalent electro-mechanical diagrams, shown in the lower part of the figure.

The two diagrams are identical except for the shunt element M_s C_s - C_c C_d and M_s C_s - C_c C_d - M_{ss} C_{ss}. The pressure in the cavity is proportional to the force f_c across C_c ($f_c = \frac{\mu_c}{j\omega C_c}$. When C_c is constant, the largest velocity u_c (and hence force f_c) is obtained when the impedance of the shunt element is small, and no diaphragm should therefore be used. Furthermore, as the diaphragm is a common shunt element for all the forces, it will not be able to give any discrimination between them and thereby influence the signal-to-noise ratio (i.e.: the ratio between f and $f_R + f_a$).

4.5.2.2. Below the resonance frequency of the skin (2-4 kc) the equivalent diagram for the coupler without diaphragm is shown in Figure A4-61.

The skin compliance C_s' under the edges of the coupler is to be interpreted as a transversal compliance that acts as a leak for C_s - C_c.

A small value for C_s' is preferable since this will give higher attenuation of f_a and at the same time give less shunting of the cavity for the velocity from the force f.

The impedance M_R - C_R has an effect similar to C_s' and should be large, indicating a high value for the compliance C_R and the mass M_R.

The compliances C_s, C_s' and C_{ss} are correlated, and we shall therefore briefly discuss the influence of the couplers' dimensions on their respective values. All three are proportional to the compliance of the skin, in this case the skin on the forehead. It is very unfortunate that the agreements between measurements of skin compliances are poor. (5, 6) Furthermore, the compliance will vary from one individual to another, and finally, it is dependent on the static pressure that is applied on the skin.

Earlier considerations (7) indicate that a value of $C' = 3 \times 10^{-5}$ m/Nt for the skin compliance under a piston of area 1 sq. cm on the forehead is applicable. (Static pressure 250 gms./cm^2)

Four parameters are necessary for evaluating the sensitivity of the coupler, namely (see Figure A4-61)

Outer diameter	n x R (meter) ($n > 1$)
Inner diameter	R (meter)
Length of cavity	h (meter)
Applied pressure on the coupler.	

The compliance C_s is simply

$$C_s = \frac{C' \times 10^{-4}}{\pi R^2} \qquad \text{(R in meters)}$$

and C_c is

$$C_c = \frac{\frac{h \times \pi R^2}{P_o \gamma}}{(\pi R^2)^2}$$

$$= \frac{h}{\gamma P_o \pi R^2}$$

Css is dependent upon the static pressure that is applied to the coupler on the forehead. This can be introduced in the calculations by adding a factor y, which is equal to 1 for a static pressure of approximately 250 gms. per sq. cm and .5 for 750 gms. sq. cm.

$$C_{ss} = y \times \frac{C' \times 10^{-4}}{\pi R^2 (n^2 - 1)}$$

It is not possible to set up a simple formula for C_s'. It must, though, be kept in mind that the wider the ring, the more it will attenuate sound transmission through it. A suitable mechanical representation is a multiple diaphragm (see Figure A4-61c). The equivalent electrical impedance is a series connection of capacitors. The wider the ring is made, the larger are the added capacitors. The resulting capacitor will, therefore, be proportional to $1/\log_e n$ instead of $1/n$.

A simplified equivalent diagram is drawn on Figure A4-62. Disregarding C_s', the pressure in the cavity is

$$p_c = f_c \times \frac{1}{\pi R^2}$$

$$= u_c \times \frac{1}{jw(C_c + C_s)} \times \frac{1}{\pi R^2}$$

$$= u_c \times \frac{1}{jw\left(C' 10^{-4} \frac{h}{\gamma P_o}\right)}$$

$$\simeq u \times \frac{1}{jw\left(C' \times 10^{-4} \frac{h}{\gamma P_o}\right)} \quad \text{for } w > w_o = \frac{1}{\sqrt{M_r C'}}$$

h must be small in order to obtain high sensitivity. A low resonance frequency is obtained by making

$$C^* = \frac{C_{ss}(C_s + C_c)}{C_{ss} + C_s + C_c} \qquad \text{large:}$$

$$C^* = \frac{1}{\dfrac{\pi R^2}{\left(C' \times 10^{-4} + \dfrac{h}{\gamma P_o}\right)} + \dfrac{\pi R^2 (n^2 - 1)}{y \times C' \times 10^{-4}}}$$

R and n should be small. Furthermore, a low resonance frequency is obtained when the static pressure on the coupler is held low ($y = 1$).

Practical values for R and h are

$R = 16$ mm

and $h = 3$ mm

inserting these values and

$C' = 3 \times 10^{-5}$ m/Nt

$\gamma P_o = 1.4 \times 10^{-5}$ Nt/m^2

gives

$$C^* = \frac{1}{\dfrac{\pi \times 256 \times 10^{-6}}{\left(3 \times 10^{-9} + \dfrac{3 \times 10^{-3}}{1.4 \times 10^{5}}\right)} + \dfrac{\pi \times 256 \times 10^{-6} \times (n^2 - 1)}{y \times 3 \times 10^{-9}}}$$

$$= \frac{10^{-3}}{33 + \dfrac{1}{y}(n^2 - 1) \times 268}$$

The smallest practical value for n is 1.2:

$$C^* = \frac{10^{-3}}{33 + \dfrac{1}{y} \times 117}$$

A pressure of 250 grs/sq. cm ($y = 1$) gives

$C^* = .65 \times 10^{-5}$ m/n$_t$

The area under the edge of the coupler is $\pi \times 1.6^2 \times (n^2 - 1) = \pi \times 1.6^2 \times .44 = 3.5$ sq. cm, and the applied force on this coupler should therefore be in the order of two pounds.

The low value of n has one drawback: decreasing attenuation under the edge of the coupler. This effect can

be partly reduced by applying a larger force, , say 5-6 pounds, on the coupler. The value for C* is thereby changed to $C^* \approx .4 \times 10^{-3}$ m/Nt and consequently the resonance frequency is shifted slightly upward.

In order to obtain further attenuation of the sound under the coupler edge, a ring with two cavities was made up of Satinflex (see Figure A4-63).

When using the coupler it is necessary to provide a release of the static pressure in front of the microphone in order to prevent a static deflection of the diaphragm. The pressure-release takes place through the acoustic filter shown on Figure A4-63.

The acoustic filter consists of three cavities with interconnecting tubes.

Tubes were chosen since they have less flow resistance than slits with same mass, i.e. tubes have higher Q. The tubes were made of aluminum, extruded over thin piano wire.

The dimensions of the cavities were chosen so as to prevent resonances below 8-10 kc. Several modes may occur in the cavities, and except for the lengthwise standing waves, are to a great extent dependent on the ratio between outer and inner diameter of the coaxial cavity. A solution of the wave equation with the boundary conditions that the radial velocity is zero for the radius equal to r_1 and r_2 (see Figure A4-64) results in the curves shown. Except for the low order tangential modes (α mo) it is seen that $\beta = \frac{R_2}{R_1}$ should

be chosen large.

For the cavities used in this filter, only the low order tangential modes α_{10}, α_{2} and α_{30} occur below 10 kc and the outer diameters of the three cavities were made slightly different so they would not resonate at the same frequencies. The resonance frequencies are shown on Figure A4-63

If the connection between two cavities consists of one tube only, the chances are great that one or more of the modes will be excited. There are, therefore, three assymetrically placed tubes between the cavities in the filter.

The lengths of the tubes were chosen as long as possible which will give the highest Q when cavity volume and cutoff frequency are given. (For the same air mass, the resistance will be smaller for a longer tube than for a shorter tube.)

The pressure in the coupler cavity will, due to the leaks, drop off at low frequencies. It was felt that the head vibrations would give high pressure levels at low frequencies and the leaks along the 640AA microphone were therefore adjusted to give a roll-off from 200 cps and down.

The attenuation vs frequency curves for the filter are shown on Figures A4-66 and A4-67.

Before the experiments with the forehead microphone are described, we shall briefly discuss how we can make an absolute measurement of the skull vibrations.

The relation between the velocity u of the skull and the pressure P_c in the cavity is, above the resonance frequency of the coupler on the forehead, given by

$$P_c = u \times \frac{1}{jw(C' \times 10^{-4} \frac{h}{\gamma P_o})}$$

The acceleration g of the skull is given by

$$|g| = w\ |u|$$
$$= w^2(C' \times 10^{-4} \frac{h}{\gamma P_o}) \times P_c$$

For the coupler used in these measurements we find

Acceleration in m/sec^2	$g = w^2 \times 25 \times 10^{-9} \times p$
Velocity in m/sec	$u = w \times 25 \times 10^{-9} \times p$
Displacement in m	$x = 25 \times 10^{-9} \times p$

where p is in Nt/m^2. If p is in dyne/cm^2 the factor 10^{-9} has to be changed to 10^{-10}.

On Figure A4-68 the results from the use of the forehead pickup is shown. Taking the leak in the coupler cavity and the resonance of the coupler on the forehead into account, we can by use of above formulae calculate acceleration, velocity and displacement of the skull.

The resonance frequency for the coupler on the forehead is

$$f_o = \frac{1}{2\pi\sqrt{M_R \frac{C^* C_R}{C^* + C_R}}}$$

When the coupler is hand-held, C_R is large, and therefore

$$f_o = \frac{1}{2\pi\sqrt{M_R C^*}}$$

MR's mass is approximately 500 grs. and C* is earlier found equal to approximately $.5 \times 10^{-5}$ m/Nt:

$$f \approx 100 \text{ cps}$$

Below 100 cps the measured pressure (Figure A4-68) must be corrected with 12 db/octave. In addition, the correction curve on Figure A4-65 must be applied. These corrections result in Curve B shown in Figure A4-68.

Applying the formula $g = w^2 \times 25 \times 10^{-10} \times p$ where g is in m/sec^2 and p is in dyne/cm^2, results in curve c. This curve shows the acceleration of the forehead in db above 1 G = 9.81 m/sec^2, and the output from an accelerometer with sensitivity 1 mV/G should therefore be the same curve, but with the other right-hand scale, giving the accelerometer output in db above μV.

4.5.3 Experimental Apparatus

4.5.3.1 Coupler
The theory leading to the design of a suitable coupler for forehead speech has been discussed in the previous section. The microphone in its coupler is shown in Figures A4-69 and A4-70-71.

4.5.3.2 Separating Plane
As was the case with the ear pickup, we desired that airborne speech not contribute to the signal picked up on the forehead. The coupler itself offered some attenuation to airborne speech, but due to the close proximity of the lips and forehead, additional shielding was required. This was accomplished by a horizontal plane which helps to increase the acoustic path from lips to forehead. Absorption was also placed under this device. This kept the airborne speech approximately 20 db less than bone-conducted speech in all bands (Figure A4-72).

4.5.4 Attenuation of forehead pickup external noise
The attenuation of external noise is shown in Figure A4-73. It was measured as the difference in sound pressure level just outside the coupler to sound pressure level inside the coupler cavity on the forehead.

4.5.5 Long time average speech spectrum on the forehead
A subject held the coupler to his forehead, and speech on the forehead and at the lips was then recorded. The same speaking

level was maintained for both recordings. These long time average speech spectra for the "joe...lawn" sentence are averaged in Figure A4-74 for several observers.

The difference between the speech level at the forehead and at the lips (rel. speech change) is shown in Figure A4-75 for the speakers. The average of this data shows a slope of -12 db/octave in the low bands, decreasing to -6 db/octave in the higher frequency bands. The -12 db/octave slope is explained from the diagram Figure A4-62. Z_B is at low frequencies a pure mass (inductance), but at higher frequencies it becomes resistive and the slope of the pressure in C_C (force over C_C) can be represented by an inductance in parallel with a larger resistance.

4.5.6 Influence of the forehead pickup on the skull vibrations

It was felt that the weight of the forehead pickup might load the head and thereby impair the skull vibrations. Two measurements of speech pickup on the forehead were therefore made, and during one of the measurements a heavy steel rod was placed on the skull. No difference in spectrum or level was observed, and it was concluded that the skin's compliance gave sufficient isolation between the skull (Z_B) and the forehead pickup (M_R). This is further indicated by the low resonance frequency (approximately 100 cps). between the forehead pickup and the underlying skin.

4.5.7 Improvement in S/N ratio when using the forehead pickup

By combining data on the attenuation of the coupler to external noise, and net speech change data, we can compute the improvement in signal-to-noise ratio when using this forehead pickup compared to the reference system, a pressure microphone in open. This is shown for 3 observers in Figure A4-76, curve 1.

4.5.8 Additional forehead pickups

4.5.8.1 640AA coupler with auxiliary aluminum diaphragm

A .025" thick aluminum diaphragm was sealed to the coupler (see Figure A4-69). Data combined shows an improvement in signal-to-noise ratio over the reference system as shown as curve 2 in Figure A4-76. No significant difference is noticable between the coupler with or without the diaphragm. This is in agreement with theory, as indicated in paragraph 4.5.2.1.

4.5.8.2 Magnetic Microphone

A small magnetic receiver unit with a circular diaphragm .007" thick was obtained. It is shown in its coupler in Figure A4-77 - 78. The improvement in signal-to-noise ratio using this device is shown as curve 3 in Figure A4-76. Results are similar to the 640AA microphone pickup. A water pillow placed between the microphone diaphragm and the forehead was tried in the interest of improved comfort due to distribution of pressure; however, the signal-to-noise ratio is decreased at low frequencies.

This magnetic unit was chosen for use in the articulation testing program since its small size permitted its use under

a helmet (Figure A4-79). In Figure A4-80 the equalized long time average speech spectrum of this microphone is compared to a pressure microphone at the lips. Similar results were obtained on other observers. The additional reduction in noise achieved when the magnetic unit is worn under the MA-1 helmet is shown in Figure A4-81. No part of the unit was allowed to touch the helmet. The improvement in signal-to-noise ratio over a pressure microphone at the lips in the open is averaged for observers TW and MG in Figure A4-82.

Appendix

4.6 Speech Pickup on the Teeth

4.6.1 Introduction

The possibility of detecting speech through vibration of the front teeth was discussed with members of the Panel of Experts. Even in our proposal before the program, we stressed the search for a pickup "region" which could maximize the ratio of consonant-to-vowel energy in speech. Since the consonant sounds are in general produced forward in the mouth, in fact in some cases, by the constricted flow of air between the tongue and the teeth, while the vowel sounds are produced by oral resonance with a more open mouth position, it was natural to suspect that tooth vibration should exhibit a larger proportion of consonant sound. Hence, this method of speech sound detection seemed most prospective from the start.

With the development of small accelerometers, it became feasible to construct a microphone which could be attached to a tooth. A tooth microphone was constructed and in this section we discuss the measurements which were made to compare the operation of this microphone to other systems.

4.6.2 Apparatus

4.6.2.1 Transducer

A Columbia Research Laboratory Model 607 Accelerometer was chosen because of its extremely small size. It has a sensitivity of 1 mv/g and has a useful range of from 5-50,000 CPS. A drawing is shown in Figure A4-83.

4.6.2.2 Mounting

One problem was the mounting of the accelerometer on a tooth. Of course, the type of mounting required would depend upon which tooth was used. Intuitively one would expect that the incisors should give highest consonant intensity since they are closest to the formation of many consonants. With this as a guide, we hand-held the accelerometer or several teeth. Although speech was somewhat impaired by the hand since it blocked the lips, speech on the upper jaw and on the front teeth sounded most natural. Vowel energy was higher on teeth of the lower jaw.

We therefore consulted with Dr. R. E. Groetzinger, West Los Angeles dentist, on mounting schemes. He individually fitted a "Rocky Mountain" tooth cap to two subjects, JPC and MG. These are simply slipped up over the tooth and held by friction at the contact points. A 4-48 machine screw nut was welded to the cap. This technique was difficult and the prospect of fitting several other subjects was not attractive.

Another scheme was tried which proved to be successful and much quicker. Small bands of .002 inch thick brass shim stock to which a 4-48 nut was soldered were constructed. These are slipped up over the tooth until good contact is made. Once it is on, the subject "forms" the band to his tooth by pressing his tongue and the teeth of the lower jaw to the back of the band while pressing the front of the band with his finger. Using this technique, a very rigid contact is assured, and the bands require only a few minutes to build. A photograph of the accelerometer, tooth bands and caps, appears in Figure A4-84.

The accelerometer is then screwed into the nut on the band or cap and the assembly is slipped up over the tooth. The accelerometer protrudes through the lips and the lead extending from it is led to one side, as shown in Figures A4-85, 86 and 87. Very little if any distortion in speech occurs because of this encumbrance. The reader may convince himself of this by pressing the eraser end of a lead pencil to his tooth and then speaking.

4.6.3 Long-Time Average Speech Spectra Using the Tooth Microphone

In Figure A4-88 speech spectra for the tooth pickup and pressure microphone at the lips are drawn. Both are at the same speaking level. The db scale is arbitrary. Instrument noise did not permit reliable measurements in some bands, and they are therefore omitted. The acceleration spectrum shape on the teeth is seen to be quite similar to that at the lips except in the 1200-2400 CPS band, which may be due to a tooth resonance.

4.6.4 Improvement in Signal-to-Noise Ratio Over an Open Pressure Microphone Using Tooth Pickup

The signal-to-noise ratio for the tooth microphone and open pressure microphone for a fixed speech level, and noise level was measured. These two S/N ratios were then compared. The result gave the improvement in signal-to-noise ratio of the tooth microphone over the open pressure microphone. This was done both in the open and with the MA-1 helmet on the subject's head. Results are shown for the three subjects in Figure A4-89. These results show the tooth microphone to be superior to any of the experimental microphone systems tested thus far on the basis of improvement in signal-to-noise ratio over the reference system.

4.6.5 Advantages of Using the Tooth Microphone

1. Good signal-to-noise ratio:

This is discussed in section 4.6.4. A by-product is minimum susceptibility of a system to acoustic feedback or singing.

2. Consonant intensity is high:

The microphone is near the area where consonants are formed, therefore consonant intensity is high.

3. The microphone is fixed to the mouth:

Since it cannot move, it will not suffer degradation in performance that others do as they move away from the mouth.

4. Relatively comfortable:

Will be more comfortable than ear or forehead microphones and less annoying than microphones placed against the lips.

4.6.6 Disadvantages of Using a Tooth Microphone

1. Physiological aspect of putting foreigh material into mouth.
2. Problem of electrical leads supported outside the lips.
3. Custom fitting may be required.

Appendix

4.7 Physical Studies and Data on AIC-10 Components

4.7.1 Summary

Three microphones used with the AIC-10 system were made available to this laboratory. They included:

1. M-32/AIC microphone which is fitted into MS22001 oxygen mask.
2. M-33/AIC microphone
3. M-34/AIC microphone in MX1334/U noise shield.

They are shown in Figures A4-90-93. A few physical measurements were performed on these systems and they were also tested in the articulation testing program.

4.7.2 Long Time Average Speech Spectra

Long time average speech spectra for the AIC/10 microphones are shown in Figure A4-94. Also shown for comparison is the LTA spectrum for the reference system, i.e. open pressure microphone at the lips.

4.7.3 Attenuation of MS:22001 Oxygen Mask and MX1334/U Noise Shield

The attenuation of the MS:22001 oxygen mask and MX1334/U noise shield is given in Figure A4-95 respectively.

4.7.4 Improvement in Signal-to-Noise Ratio of AIC-10 Microphones

The improvement in signal-to-noise ratio of the AIC-10 microphones relative to the reference system is shown in Figure A4-96.

4.7.5 Response of the M-33/AIC microphone as a function of distance from the lips.

The decrease in response of the M-33/AIC microphone as a function of distance from the lips is shown in Figure A4-97.

4.7.6 Improvement in Signal-to-Noise Ratio as a function of distance from the lips.

Combining the results of Figures A4-10, A4-96 and A4-97, the improvement in signal-to-noise ratio of the M-33 microphone as a function of distance from the lips is obtained and shown in Figure A4-98. The microphone performance decreases as it moves from the lips although it is still effective 1 inch away. If the microphone is moved above or below the lip line, its performance also suffers.

Appendix

4.8 Evaluation of Transducers on the Basis of Articulation Testing

4.8.1 Summary

A new scheme has been devised to measure the relative effectiveness of communication systems in noise.

Special word lists were constructed and a shortened testing procedure was devised.

Each of the microphone systems described in Sections 4.2 - 4.7 were tested and are rank ordered on the basis of intelligibility. An estimate of the maximum noise field in which each will yield a given intelligibility for a fixed speech effort is given.

4.8.2 Introduction

Early in the contract, after conferences with members of the Panel of Experts, it became apparent that for our own purpose the process of articulation testing needed critical review. Three major questions to be considered were:

(1) Can word lists be optimized to provide greater efficiency in terms of information per unit time?

(2) Can a procedure be devised to effectively rank order microphone systems?

(3) Can an efficient procedure be found which will furnish not only accurate relative performance data, but also will be absolute with sense that the ultimate possibilities of a given system can be determined in terms of vocal effort versus ambient noise level.

With the help of the Panel these questions were explored. Optimized word lists, techniques for rank ordering systems and efficient procedures were developed. These are discussed in detail below.

4.8.3 Word Lists

4.8.3.1 Word Form

On the basis of conferences with members of the Panel of Experts and members of the Haskins Laboratory and other cooperating groups, it was decided to use consonant-vowel-consonant (cvc) syllables as the test material for use in intelligibility testing. CVC words have several advantages over other material:

(1) Experience with articulation testing has shown that vowel sounds are relatively durable, i.e. resistant to noise in interference as compared with consonant sounds. Hence, insofar as vowels contribute to recognition in words, they dilute the sensitivity of the test. Therefore, using the most difficult consonant sounds in the word lists should give the most sensitive test and the greatest discrimination between systems.

(2) The use of familiar English words tends to increase recognition and hence dilutes the sensitivity of the tests. In the case of PB words, familiarity with the words is assumed so that a learning period is required. If the

words are used in fixed word lists, another memory factor is introduced. In contrast, when using the CVC words the intelligibility test becomes more purely a recognition task. Hence, it has been found that an extensive learning period is not required. Fixed lists can be eliminated by card shuffling.

4.8.3.2 Choice of Consonants

An analysis of the article by Miller and Nicely was made in order to choose consonants for the CVC words which would be difficult to perceive in noises. From the confusion matrices given in the article, we have determined the percentage of times that a particular consonant was confused under a certain S/N ratio or bandwidth. This analysis is shown in Table A4-IIa. A summary is shown in Table A4-IIb.

Since a bandwidth of at least 200-6500 CPS was anticipated, the results at this bandwidth were incorporated to rank order the consonants as shown in Table A4-III.

On the basis of these results 14 consonants were chosen and are listed in Table A4-IV.

4.8.3.3 Choice of Vowels

At first it was hoped that only one vowel could be used to shorten the word lists. However, we were advised by the Panel that the use of only one vowel in CVC words is too limited because the formant transitions which contribute to intelligibility behave differently for a given consonant with different vowels.

Three vowels were chosen:

a, as in father

u, as in shoe

i, as in hit.

The choice of vowels was based on Fletcher's Frequency of Occurance Chart . They also give a good representation of front, mid, and back vowels.

4.8.3.4 Formation of CVC Words

The 14 consonants and 3 vowels were used in all possible combinations forming a total of 578 words which were written on 3 x 5" cards. Approximately 50 words were removed because they were difficult to pronounce, or vulgarisms. These cards are re-shuffled after use, so that there are no fixed lists.

4.8.3.5 Carrier Sentence

The CVC words were imbedded in the carrier sentence, "Write the word ___ in the blank." This helps the talker to speak the word more naturally and to permit influence of both a prior and later word upon recognition. The word "blank" in the carrier sentence was stressed rather than the test item. The intonation pattern was therefore ....

4.8.4 Articulation Testing Crew

Three students from UCLA who majored in speech were engaged. They were selected because we felt their interest would be better sustained throughout the program than those whose interest lay in another field.

4.8.5 Articulation Score of CVC words versus S/N Ratio

The articulation score as a function of S/N ratio for the 500-odd CVC words described in Section 4.8.3 was measured. The results are shown in Figure A4-100. The articulation score is determined by the total number of consonants heard correctly divided by the total number of consonants in the list x 100%.

The purpose of the test was to provide a function of percent articulation versus S/N which could be used later for extrapolation to find the value of S/N which would have given 50% (or any other nearby value), even though a particular test had resulted in some other value of articulation. Incidentally, the placement of this test in the program also provided a short training period for the crew.

The tests were conducted in the following manner. The 500 words, each in a carrier sentence, "Write the word _____ in the blank" were recorded on an Ampex 350 tape recorder. Three speakers, TW, WO and MG, each recorded approximately the same number of words. The words were divided into lists of 50. The pre-recorded tapes were then played back and mixed with a jet noise spectrum at various values of S/N ratio. A block diagram of the apparatus is shown in Figure A4-99.

The words were recorded in a soundproof room with highly absorptive walls. The microphone was 12 inches from the subject's lips and the voice level was monitored on a damped meter. The subject was told to gage his level by the carrier sentence and not to stress the test item. A level of 70 db SPL at 1 ft was chosen arbitrarily.

When the recordings were played back the long-time average of the carrier sentence determined the level of the signal. The long time average of the noise level was also used. The output level of the tape recorder remained constant, while the noise level was changed by means of the attenuator.

The three speakers, TW, WO and MG, were also used as listeners. The choice of talker and the S/N ratio were made at random. The tests ranged from 8 db S/N to -16 db S/N. Approximately 20 tests of 50 CVC words were conducted at each level.

4.8.6 Parameter Used in Rating Microphone Systems on the Basis of Articulation Testing Results

The following scheme was devised to measure the relative

effectiveness of communication systems in noise: to measure the talking effort level relative to maximum vocal effort which is required to produce a given articulation percentage in a given noise field. The given articulation percentage chosen was 50%, a point at which maximum sensitivity occurs and there is the most room for variation among articulation score obtained with different communication systems.

Maximum effort is defined in the list of definitions. This effort was chosen as a stable reference point for a given speaker, which has several advantages. For example, some systems which impair speech, such as a microphone in an oxygen mask, will lower the maximum speech effort attainable. This should be taken into account since speech encumbrance will reduce a microphone's effectiveness at any talking effort. Such a limitation will show up in this testing procedure. Maximum effort is also, in our opinion, more stable from day to day than any other reference level, such as "normal conversational level," and is independent of a subject's ability to hear his own speech, as for example when a helmet covers his ears or the noise field masks his speech.

The general approach of changing the speaking effort at a fixed noise level rather than changing the noise level at a fixed speech effort to achieve a 50% articulation score, was followed for the following reason: for many systems to be tested, noise on the order of 130-140 db would be necessary to reduce the scores to 50% at nominal speaking efforts. Since the noise levels in WADC TN 56-411 indicated a sound envelope for jet aircraft in level flight of less than 120 db overall, an increase to 130-140 db seemed unrealistic.

The approach of varying the speaking effort also follows from actual conditions wherein a talker will raise his voice until he is understood, and will generally not talk at a higher level than is necessary.

4.8.7 Monitoring Speech in Noise Fields at Negative S/N Ratios

Briefly stated, the problem arose as to how to monitor speech at low S/N ratios during the articulation testing. Low S/N ratios were dictated by the requirement that the articulation scores be approximately 50%. With some systems this meant a S/N ratio of -10 db. No monitor system was available which under the same conditions would give a positive S/N ratio and thus would allow the subject to monitor his speaking level.

Several approaches were considered. These included (1) a method whereby static pressure generated in the vocal cavities would be monitored and correlated to speech effort, (2) Electromyography - electrical potentials generated in the chest muscles

during speech would be correlated to speech effort. Both of these ideas were abandoned in favor of a much more simple technique which was suggested by Dr. Gordon Peterson. He pointed out that it is possible for some persons to learn to talk at a given level in the quiet, and, when noise is introduced, to talk at exactly the same level. He suggested that it was a matter of practice to acquire this skill.

Hence, we undertook this technique as a method of maintaining a given speech level when the S/N ratio is negative. A program to acquire this skill was begun. The testing crew took turns reading CVC words at a given level into a microphone in the quiet. This was monitored by a well damped VU/meter which each crew member could see. Noise was then injected into his ears through headphones and the VU/meter was hidden from his view. Other members of the crew determined by watching the VU/meter if his level had changed. After approximately a week of practice each crew member, after establishing a level in the quiet, could hold this same level within 1 db over a list of 25 words after the noise was introduced. Various speech levels were attempted to simulate the level which would be required in the actual testing program. Tests of this type were also conducted using external noise in the noise enclosure with the same repeatability. After this practice we felt reasonably sure that the crew could maintain any given level during the test over a list of 25 words with a little practice before each test.

4.8.8 Description of Apparatus & Articulation Testing Apparatus

A block diagram of the apparatus is shown in Figure A4-101 The function of the components will be explained in the procedure given below.

Procedure for running articulation test:

1. With the subject seated in the noise enclosure a 120 db jet noise spectrum is introduced. With no speaking, the noise as picked up by the microphone system under test is read on the voltmeter. Let this noise reading be N db.

2. The noise is then turned off. The speaker then begins reading the list of CVC words (these words are not used in subsequent tests) placed in the carrier sentence "Write the word ___ in the blank." He has been instructed not to stress the test item but to allow it to fall where it may in sentence. His average speaking level is monitored by a VU/meter which has been damped to get an average reading for the sentence. Zero VU is the target point. It is also monitored by the listeners on the voltmeter which has also been damped to obtain an average level. Let this average speech level in the quiet be S db as read on the meter. Hence, a static signal-to-noise ratio - S/N in db is established.

3. In the test we were striving for an articulation score of approximately 50%. Since the noise level is fixed, the score will be a function of the speech level. The listeners can control the talker's speaking level

by means of the attenuator. For example, if 2 db of attenuation is added, the talker must speak on the average of 2 db louder to make the VU/meter read zero VU.

4. Since we do not know what value of S/N will give an articulation score of 50% with each system, a guess is made. After the talker establishes his speaking level for this value of S/N (usually requiring about 5 - 10 words) his VU/meter is turned off. The listener's voltmeter, however, still monitors his speech and they determine if his level has shifted (usually requires about 5 words). If his speech level remains at the given level, the noise is turned on and the test begins.

5. The talker reads 25 words; the listeners write down what they hear, the noise is turned off and the listeners recheck the talker's speaking level. If it is within 2 db of the required level they continue, and the talker reads 25 more words. The talker then comes out and the scores are checked.

6. If the articulation score (number of consonants right divided by the total number of consonants x 100%) differs considerably from 50%, the test is voided and a new speaking level is chosen. The procedure is then repeated until scores of approximately 50% are obtained.

7. Several tests are then taken at this value of S/N using each of the crew of 3 as a talker. The number of tests for each system varied. This procedure proved to be quite efficient once the articulation crew became familiar with it.

4.8.9 Results of Articulation Testing

The results of the articulation testing program are shown in Table A4-V, averaged for the three talkers. The column headed "DB Below Max. for 50%" is obtained by extrapolation using the information contained in Figure A4-100. For example, suppose microphone "A" gave an articulation score of 65% at a speaking effort 30 db below maximum speaking effort. We enter the figure at a score of 65% and find it corresponds to a S/N = - db. A 50% score corresponds to S/N = -12 db. Hence, if the talker spoke at 30 db (12-9) = 33 db below maximum, a 50% score should have resulted.

The rank order of the microphone systems on the basis of db below maximum effort for a 50% CVC score is given in Table A4-VI, column 1. The results have been averaged for the three talkers.

4.8.10 Speaking Effort Required to Achieve a 50% CVC Score at Various Speaking Efforts

The results of Sections 4.8.8 and 4.8.9 are given relative

to maximum speaking effort. It is obvious that maximum speaking effort can be maintained for only a short duration. Therefore, we would like to know the results which will arise for other speaking efforts. With the aid of previous results we will find the overall level of the jet noise environment which will yield a 50% CVC score for one particular speaking effort - "raised" and may be extended to other speaking efforts.

Sound pressure level versus subjective effort was discussed in Section 4.2.3.3 (see Figure A4-9). The difference in decibels maximum effort and the other speech efforts is tabulated in Table A4-VII.

The procedure for determining maximum overall jet noise level for a 50% CVC score is outlined in Table A4-VI, columns 2 and 3. Column 2 is 27 db (i.e. the difference between raised and maximum efforts) subtracted from column 1. Column 3 is the result of column 2 subtracted from 120 db jet noise. Column 3 is the maximum overall jet noise field for which a 50% CVC score may be achieved at a raised speaking effort.

This technique is valid if one assumes that a given S/N ratio is required for . given intelligibility. Hence, if a speaking effort is decreased by X db, then the noise must also be decreased by X db for the same intelligibility.

The results of this evaluation are shown in Figure A4-10a. The results may be generalized to other speaking efforts with the help of Table A4-VII and are shown in Figure 13.

Appendix

4.9 Evaluation of Speech Sound Alteration, Quality and Speaker Recognizability for Microphone Systems Tested.

4.9.1 Description of Sample Recordings

In order to evaluate each microphone system subjectively as to its speech sound alteration, quality and speaker recognizability, a series of tape recordings were made. These recordings consisted of the following material for each microphone system:

(1) "Joe - - - lawn" sentence, into a Western Electric 640AA microphone 4 inches from the lips. The quality of other systems is compared to it.

(2) "Joe - - - lawn" sentence into microphone under consideration.

(3) About one paragraph of reading material from "Fortune" magazine, chosen at random.

(4) Ten CVC words in the carrier sentence "Write the word __ in the blank."

(5) Ten CVC words were read in isolation, e.g. fas, siv, etc.

(6) The sentence "Sister Susie sells silk shirts by the sea shore."

(7) The sentence "Peter Piper picked a peck of pickled peppers."

(8) Five Air Force landing commands.

These recordings were made in the quiet and in 120 db jet noise with the same speaking effort (raised voice) for all systems.

The choice of material allowed us to evaluate all speech sounds in connected speech, isolation, and to concentrate on sibillants and plosives.

4.9.2 Evaluation Procedures and Definitions

Speech Sound Alteration: The four general classifications of speech sounds, i.e. vowels, fricatives, plosives and nasals, were evaluated separately. The following functional definitions were required:

(1) Natural - indicates there is little or no change in the intensity, frequency or duration of the sound as reproduced by the microphone as compared to the Western Electric 640AA four inches from the lips (the standard).

(2) Accentuated - indicates the strengthening of a particular speech sound in intensity which may be a function of frequency.

(3) Diminished - indicates the decrease in intensity of a particular speech sound. The decrease in intensity may be a function of frequency.

4. Altered - indicates that a speech sound has changed quality to the extent that it is perceived as another sound.

Quality: An attempt was made to rate the overall quality of the speech, not concentrating on any classification as in the above. The following functional definitions were used:

(1) Natural - same as above.

(2) Nasal - the quality of speech sounds when the nasal cavity is used as a resonator; especially when there is too much nasal resonance.

(3) De-nasal - the absence of resonance in the nose; the voice sounds as though "the talker has a cold in the nose."

(4) Oral - nasal resonance is normal, but the quality is as if the oral aperture is extremely large creating a dull mushy sound; "he talks as if he has a potato in his mouth."

(5) Metallic - a spread of energy at high frequency, producing noisy, discordant elements.

Recognizability: The ability to recognize the voice of the talker was rated as good, fair, or poor.

4.9.3 Result of Evaluation

Each microphone system was evaluated with respect to speech sound alteration, quality and recognizability. The tapes were played back on a high quality system into a W.E 755 loudspeaker. Two listeners took part, one a student with experience in speech therapy and the other a physicist. The results of the evaluation are shown in Table A4-XIII It should be stressed that the evaluation was completely subjective since no attempt was made to determine the physical properties of the speech with instruments.

Appendix

4.10 Speech Sound Analysis

4.10.1 Summary

Early in our testing program an attempt was made to correlate physical measurements of consonant-to-vowel and consonant-to-noise ratios with articulation testing in order to determine the relative performance of communication systems.

Some measurements of this nature were made; however, little correlation existed between the physical and articulation tests. On the basis of these results, several articles in the literature, and a conference with personnel at Haskins Laboratory, this approach was abandoned.

This appendix contains some examples of the speech analysis employed to determine consonant-to-noise ratio, and some discussion the reasons for abandoning this technique of evaluating microphone systems.

4.10.2 Relative level and frequency distribution of certain speech sounds

4.10.2.1 Instrumentation and procedure

A block diagram of the apparatus is shown in Figure A4-102. Spoken material was recorded on magnetic tape. Selected words and phrases were cut out and spliced into loops so that the material to be inspected would occur periodically.

The time constant of the rectifier filter used was .01 second and the logarithmic attenuator had a range of 30 db. A keying pulse recorded on tape provided a signal to trigger the sweep circuit of the oscilloscope.

Using a long persistence oscillograph tube, the envelopes of words remained visible for several seconds in a darkened room. The recurrent signal provided by the tape loop restored the image regularly, allowing detailed inspection of the structure of these speech sounds.

4.10.2.2 Analysis of "Joe . . . lawn" sentence

The sentence "Joe took father's shoe bench out, she was waiting at my lawn," was first analyzed. The talker spoke at normal conversational level into a 640AA microphone 18 inches in front of his lips. Included also is the analysis of a set of CVC (consonant-vowel-consonant) words consisting of fricative consonants and the vowel "a" as in father. The results of these investigations are shown in Figures A4-103, A4-104 and A4-105.

4.10.2.3 Spectra of some CVC words

The spectra of all the consonants and vowels from two sets of CVC words were also measured and are plotted in Figures A4-106 and A4-107. These results agree essentially with those given by Fletcher ("Speech & Hearing in Communication" 1953) and OSRD Report #3106, and hence furnish a cross-check on our procedure.

A comparison was made between the spectra of the set of CVC words using "a" (see above) when spoken normally and when whispered. From the results of our investigation on this rather special vocabulary, it was found that the spectra or level of the consonants were not changed when the word was whispered.

4.10.3 Fricative consonant-to-noise ratio for two microphone systems, pressure and pressure gradient

In order to measure the consonant-to-noise ratio for the fricative consonants spoken in a masking noise environment of arbitrary level and shape, an indirect procedure was used. Three separate ratios were measured and combined to produce a consonant-to-noise ratio for both pressure and pressure gradient microphones as a function of frequency. (These microphones are discussed in Appendix 4.2 and 4.3 respectively.)

The results of these measurements indicated the gradient microphone gave approximately 10 - 30 db greater consonant-to-noise ratio below the 600 - 1200 cps octave band than the pressure microphone. Above the 600 - 1200 cps band the consonant-to-noise ratio of both systems were similar.

4.10.4 Comparison of articulation testing and the physical measurement of Section 5.0.3

A series of articulation tests was performed in order to determine whether there was a correlation between the physical measurements described above and the articulation score.

It was found that there was little correlation between articulation testing scores and physical measurements of consonant-to-noise ratio. This lack of correlation led us to consult with the Panel of Experts and others to explain this inconsistency.

4.10.5 Application of some basic research on speech to our attempt at correlating physical measurements and articulation testing

Numerous clues had led us to suspect that work being done at the Haskins Laboratory would be applicable to our communications work. The paper by A. Liberman, clinched our resolve to visit them. A companion article by Halle, Hughes, and Radley[12] also seemed to confirm the implications which we tended to draw in regard to our own program. Accordingly, Messrs. Veneklasen and Snow conferred with Drs. Liberman and Cooper on 22 May 1957, seeking their advice on several questions, especially:

Is there a simple physical parameter such as possibly speech-to-noise ratio or consonant-to-noise ratio, which should correlate with consonant articulation?

The following is a record of many answers to more specific questions which bear on these generalities:

1. It is probable that the measurement of S/N for the high frequency bands will not be a valid measurement of consonant articulation even for the fricative consonants.

2. Recognition even of the fricative consonants depends greatly on formant transition as well as the noise bursts or affrication. It seems likely that, in an unfavorable noise environment, the affrication clue may be lost in the noise before the formant transition clue. Hence, to measure affrication alone is probably to concentrate on a superficial factor.

3. It is quite to be expected that crucial aspects of consonant articulation may be lost in an unusual pickup system such as ear or head location.

4. The complete physical analysis which might be expected to correlate with consonant articulation would be very complex indeed, involving detailed analysis of sound spectrograms. This is far beyond the scope of this project. However, we were urged to record word lists from each representative speech system in case such analysis should be desirable at a later time.

On the basis of these answers and comments we discontinued our approach, i.e. measuring consonant-to-noise ratio. Henceforth we relied on the more simple approach of measuring overall sentence signal-to-noise ratio as a diagnostic evaluation of microphone systems. This is described in Appendix 4. 1.

4.11 Basic Speech Data

Some basic material was gathered from the literature and summarized in chart form. These charts bring together information from several sources and provide a means of obtaining information quickly.

The Charts of tables A4-1X and A4-X list the vowels and consonants with their phonetic symbols, classification, frequency of occurance, phonetic power, and confusion in noise.

In table A4-X1, the chart summarizes the vowels and semi vowels, showing the vowel diagram, lip and tongue positicns and formant frequencies.

This material has aided the personel working on this contract at WEAL especially in the choice of consonants for the CVC words.

REFERENCES FOR APPENDIX 4

1. Kryter and Pickett, "Prediction of Speech Intelligibility in Noise" AFCRC - TR - 55 - 4

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4. Moser, Dreher, and Oyer, "The Relative Intensity of S und at Various Anatomical Locations on the Head and Neck during the phonation of vowel sounds," AFRC TN 55-70.

5. E. Barney, "A Contribution to the Physiology of Bone Conduction," Acta Otolaryngolocia, 1938 (supplement)

6. E. K. Franke, "Impedance of the Human Mastoid," JASA 24, 410, (1952)

7. E. K. Franke, "The Response of the Human Skull to Mechanical Vibrations," WADC TN 54-24 (1954)

8. Miller and Nicely, JASA 27, 342, 1955

9. H. Fletcher, "Speech and Hearing in Communication," D. Van Nostrand, N. Y. 1953

10. J. P. Egan, "Articulation Testing Methods II" OSRD Report 3802

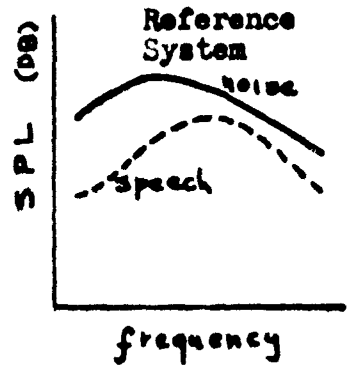
11. Liberman, JASA, 29, 117, (1957)

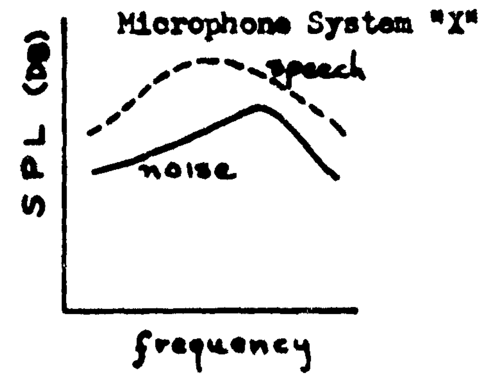
12. Halle, Hughes, and Radely, JASA, 29, 107, (1957)

PROCEDURE FOR CALCULATING THE IMPROVEMENT IN SIGNAL TO NOISE RATIO

OF A MICROPHONE SYSTEM RELATIVE TO THE REFERENCE SYSTEM.

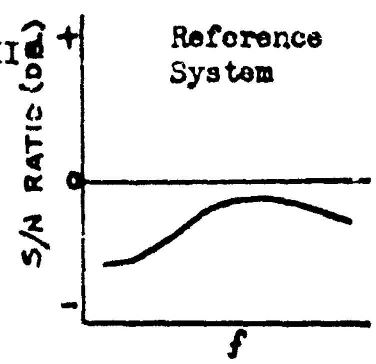
Step I

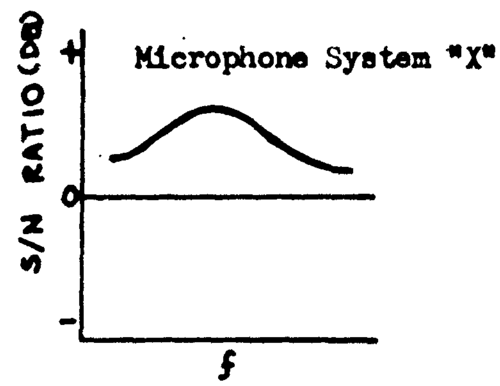




The r.m.s. noise spectrum and LTA speech spectrum are measured for the reference system and system "X". The same noise level (120 db. jet noise) and speaking effort must be used

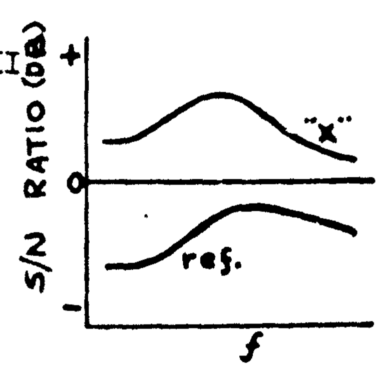
Step II





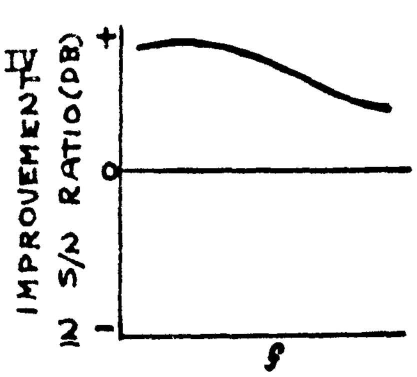
From the results of Step I the S/N ratio for each system is computed.

Step III



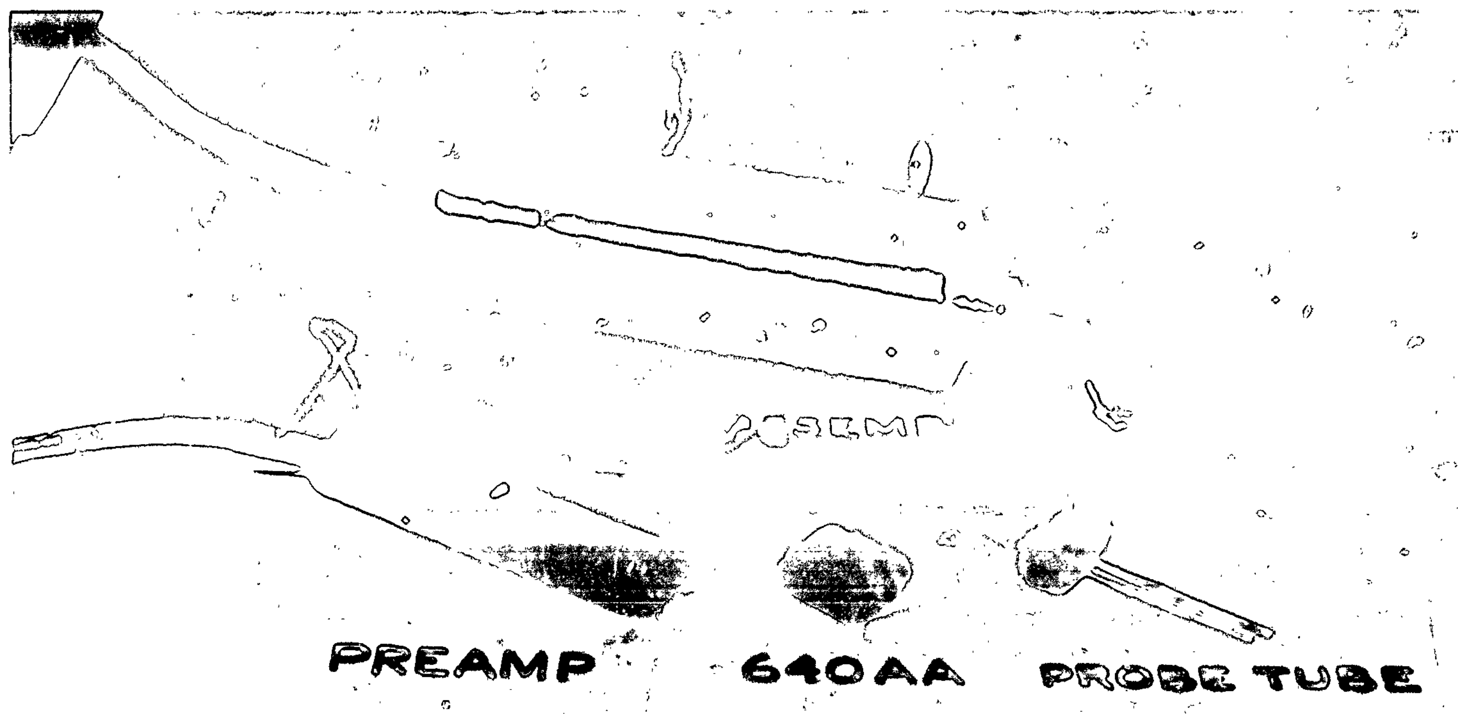
From the results of Step II the S/N ratio for each system is compared.

Step IV

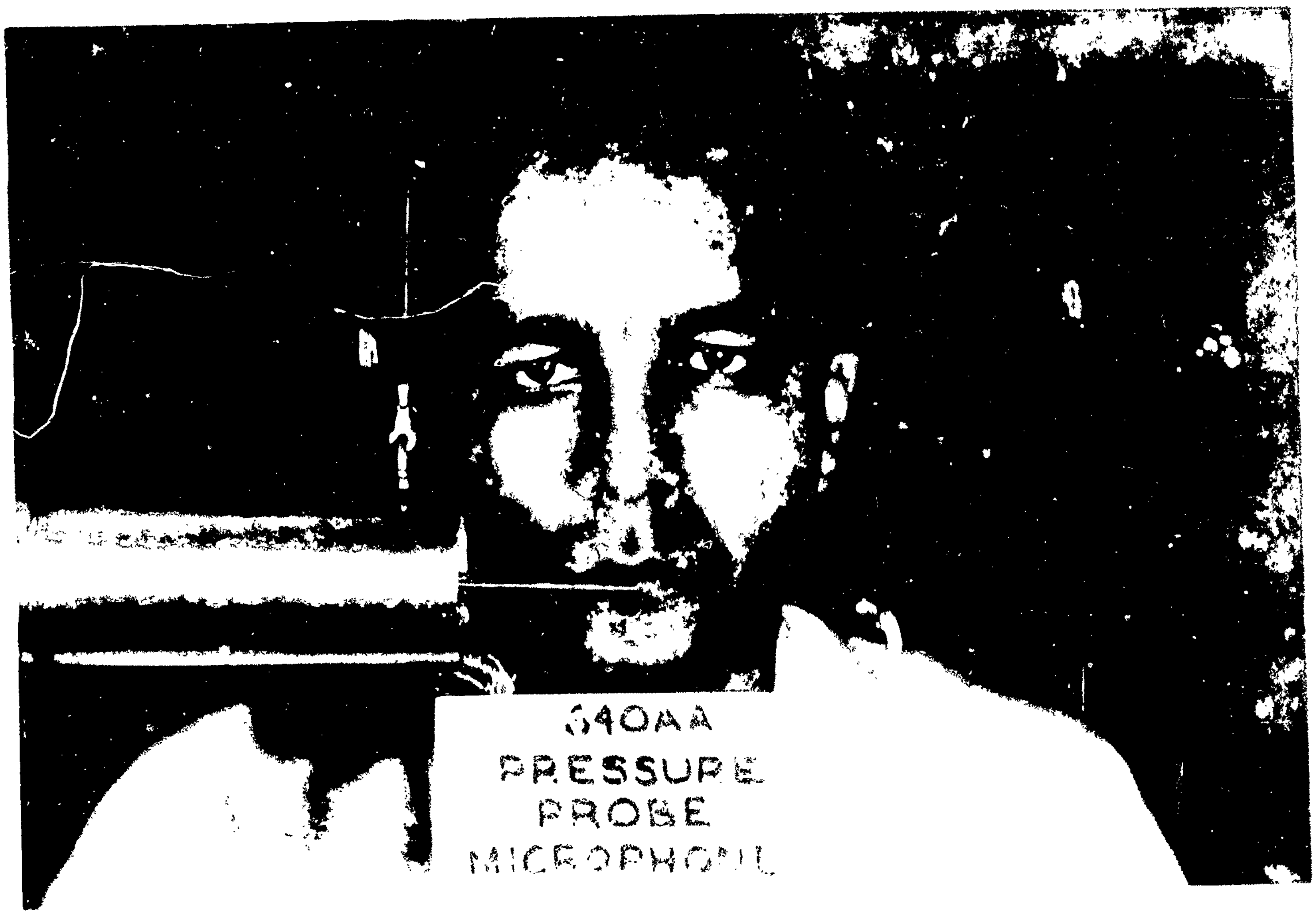


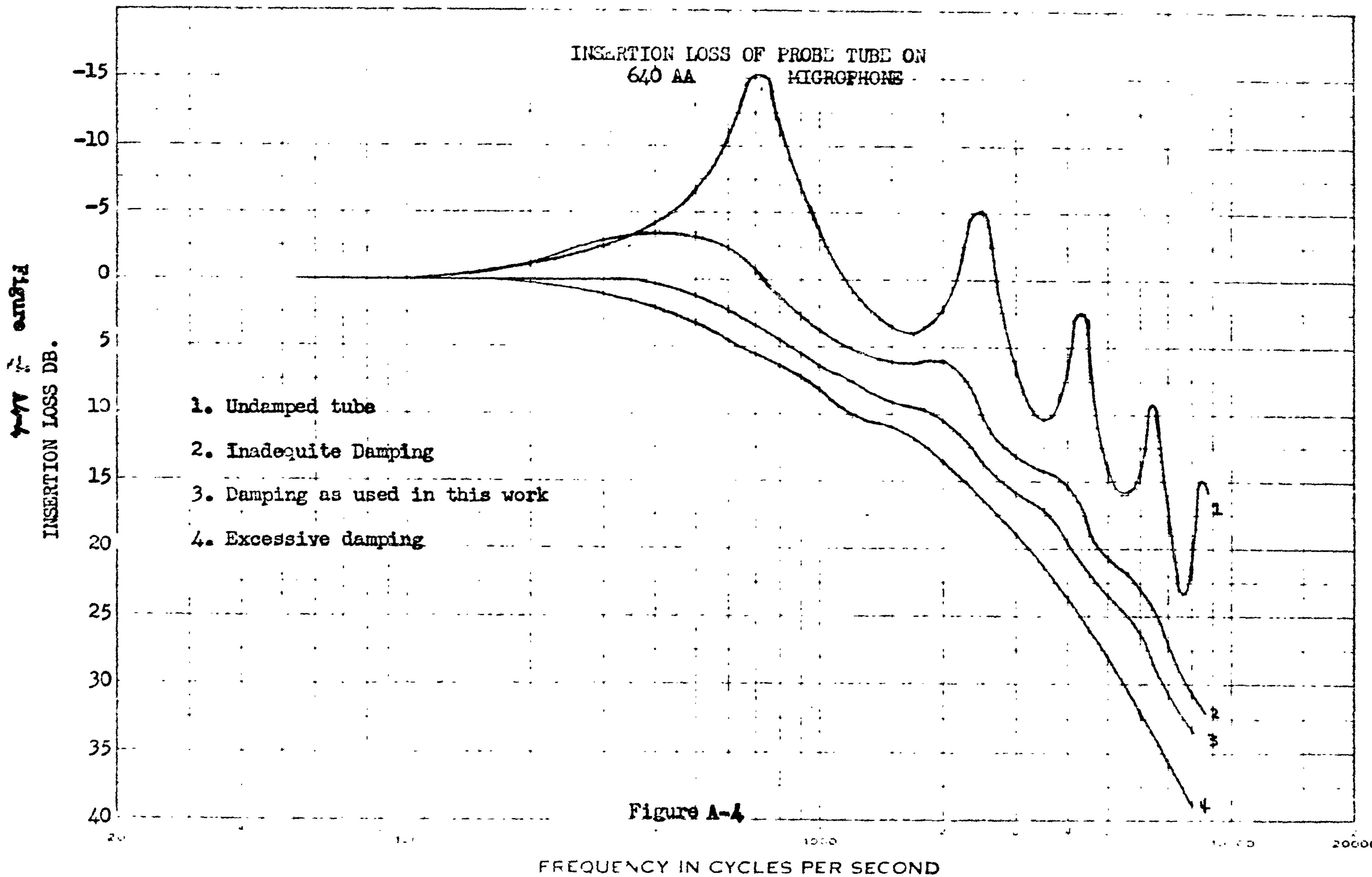
From step III the difference is S/N ratio between the two systems determines the improvement in signal to noise ratio of system "X" relative to the reference system.

Figure A4-1



PROBE TUBE
MICROPHONE


640AA
PRESSURE
PROBE
MICROPHONE



INSERTION LOSS OF PROBE TUBE ON
640 AA MICROPHONE
-15
-10
-5
0
5
10
15
20
25
30
35
40
INSERTION LOSS DB.
1. Undamped tube
2. Inadequite Damping
3. Damping as used in this work
4. Excessive damping
1
2
3
4
1000
20000
FREQUENCY IN CYCLES PER SECOND

Figure A-4

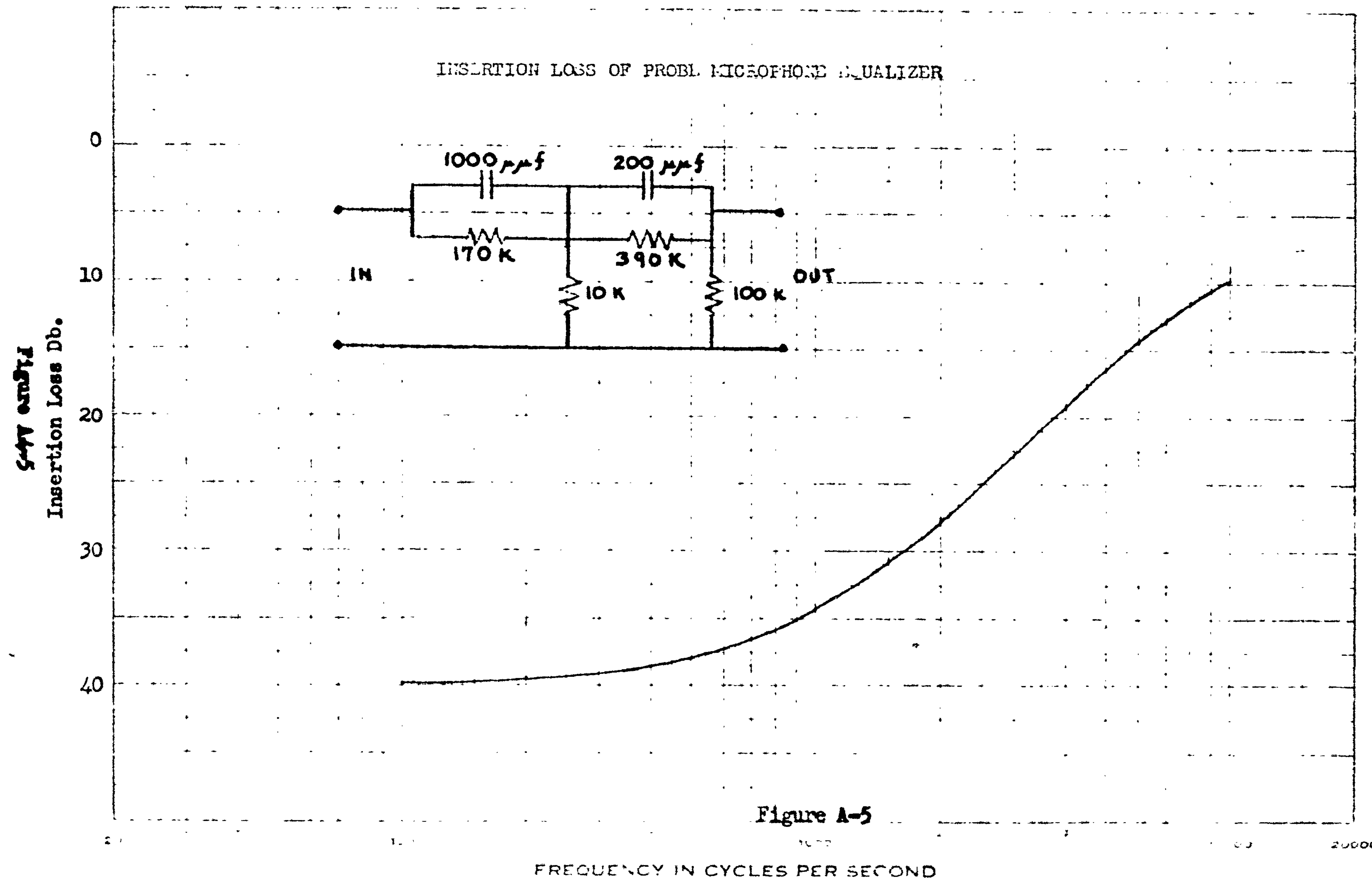
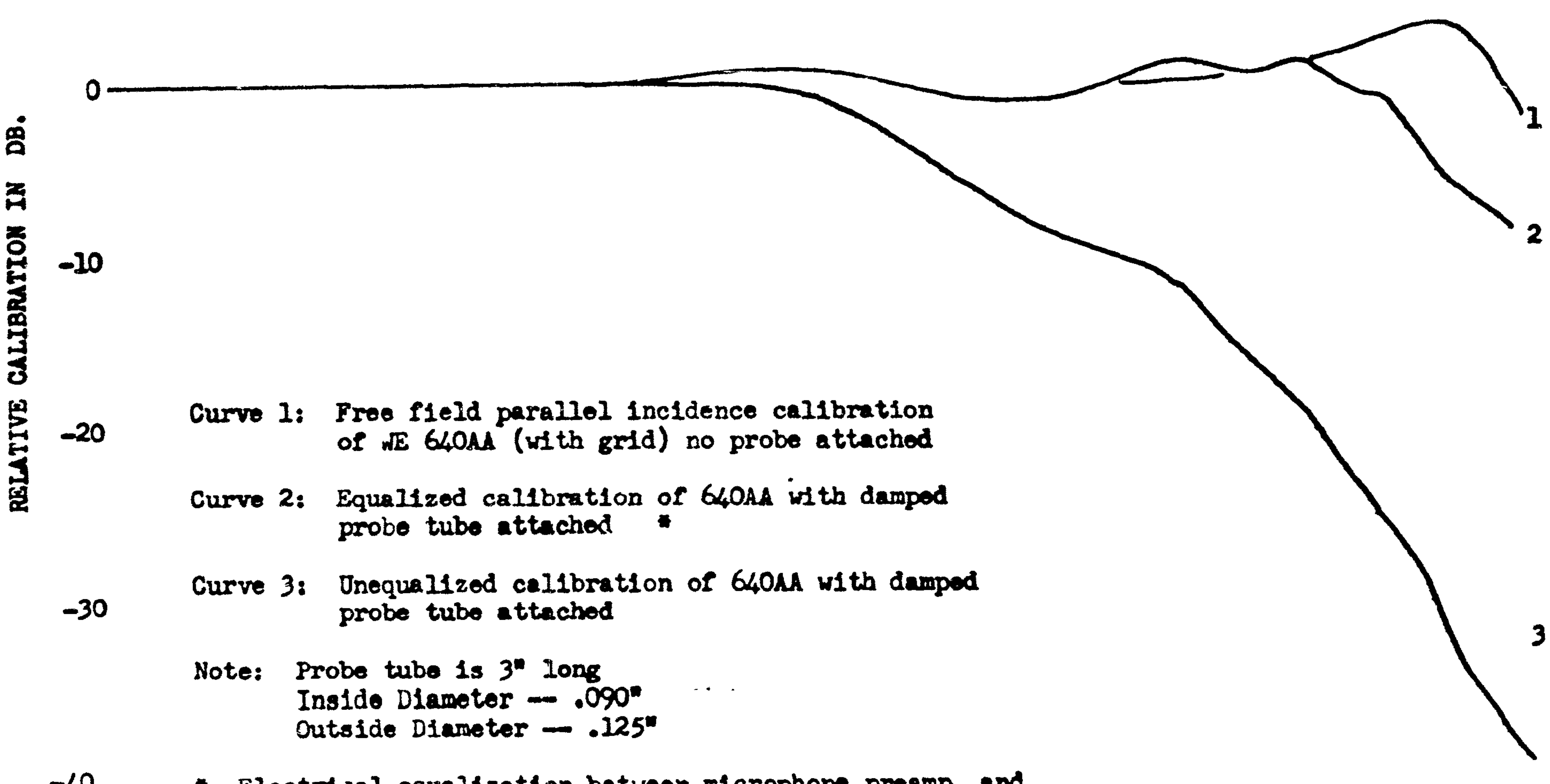


Figure A-5

Figure A4-5

RELATIVE CALIBRATION OF 640AA PROBE TUBE MICROPHONE



Curve 1: Free field parallel incidence calibration of WE 640AA (with grid) no probe attached

Curve 2: Equalized calibration of 640AA with damped probe tube attached *

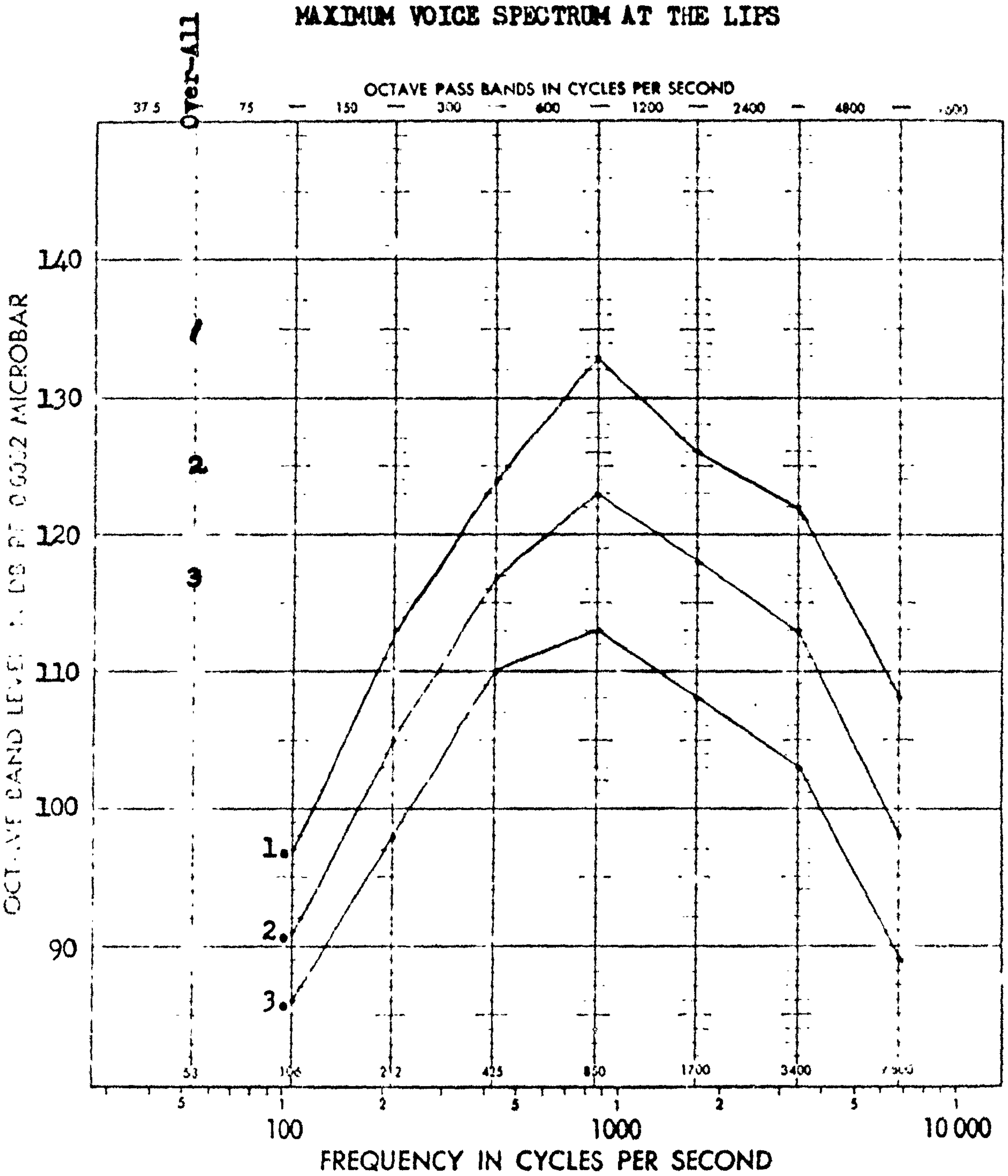
Curve 3: Unequalized calibration of 640AA with damped probe tube attached

Note: Probe tube is 3" long
Inside Diameter — .090"
Outside Diameter — .125"

* Electrical equalization between microphone preamp. and input of line amplifier introduces an insertion loss of 40 db. Therefore the absolute level of curve 2 is 40 db. less than curve 1 and 3

Figure A4-6

MAXIMUM VOICE SPECTRUM AT THE LIPS

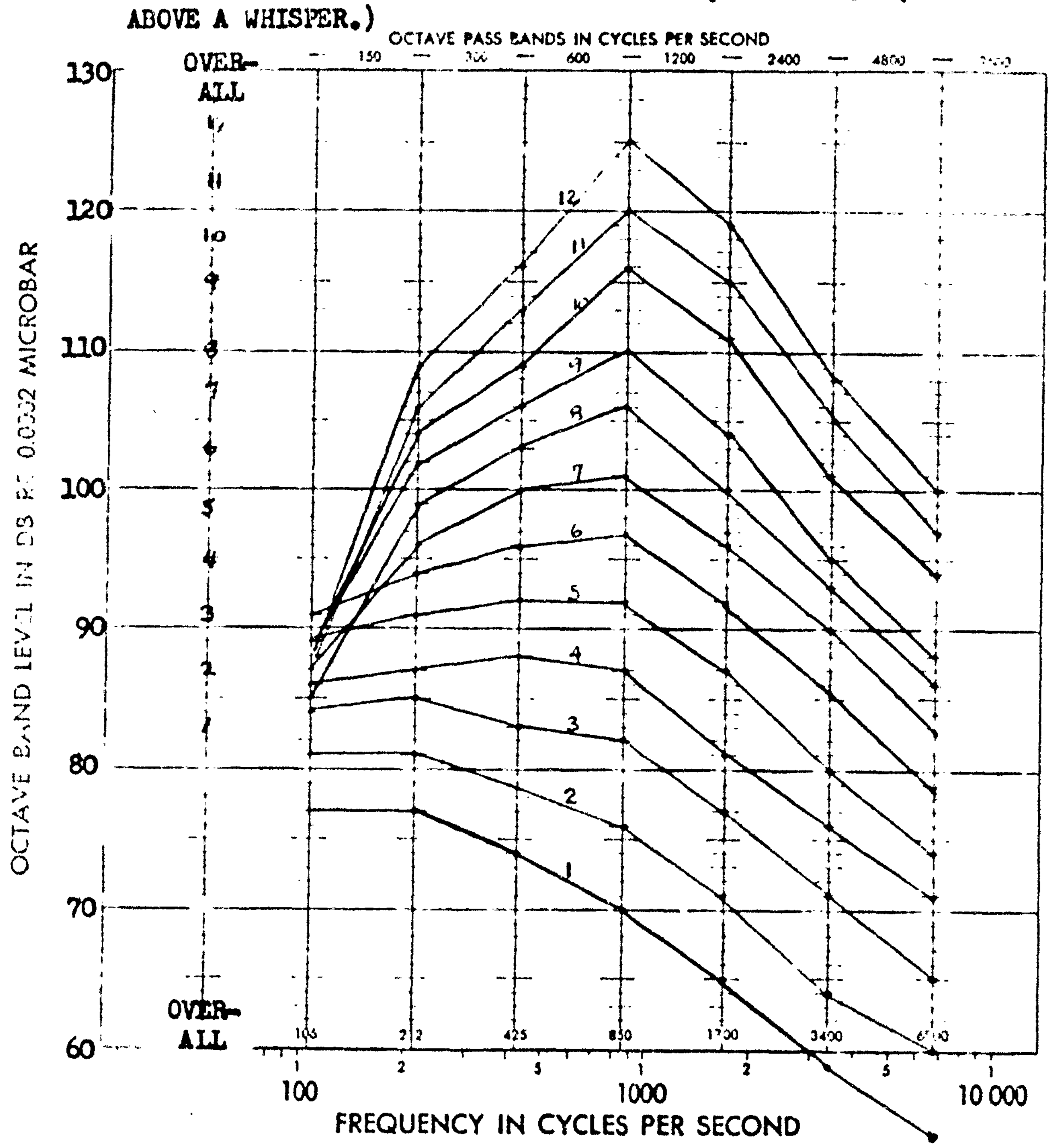


Curve 1: Highest
Curve 2: Average of 8 speakers
Curve 3: Lowest

Note: 640AA microphone was employed slightly above the lips and as close to face as possible

Figure A4-7

LONG TIME AVERAGE SPEECH SPECTRUM AT THE LIPS FOR JOE...LAWN SENTENCE. THE OVER-ALL LEVEL DECREASES IN APPROXIMATELY 4 DB INCREMENTS FROM MAXIMUM EFFORT TO VERY QUIET SPEECH (JUST ABOVE A WHISPER.)



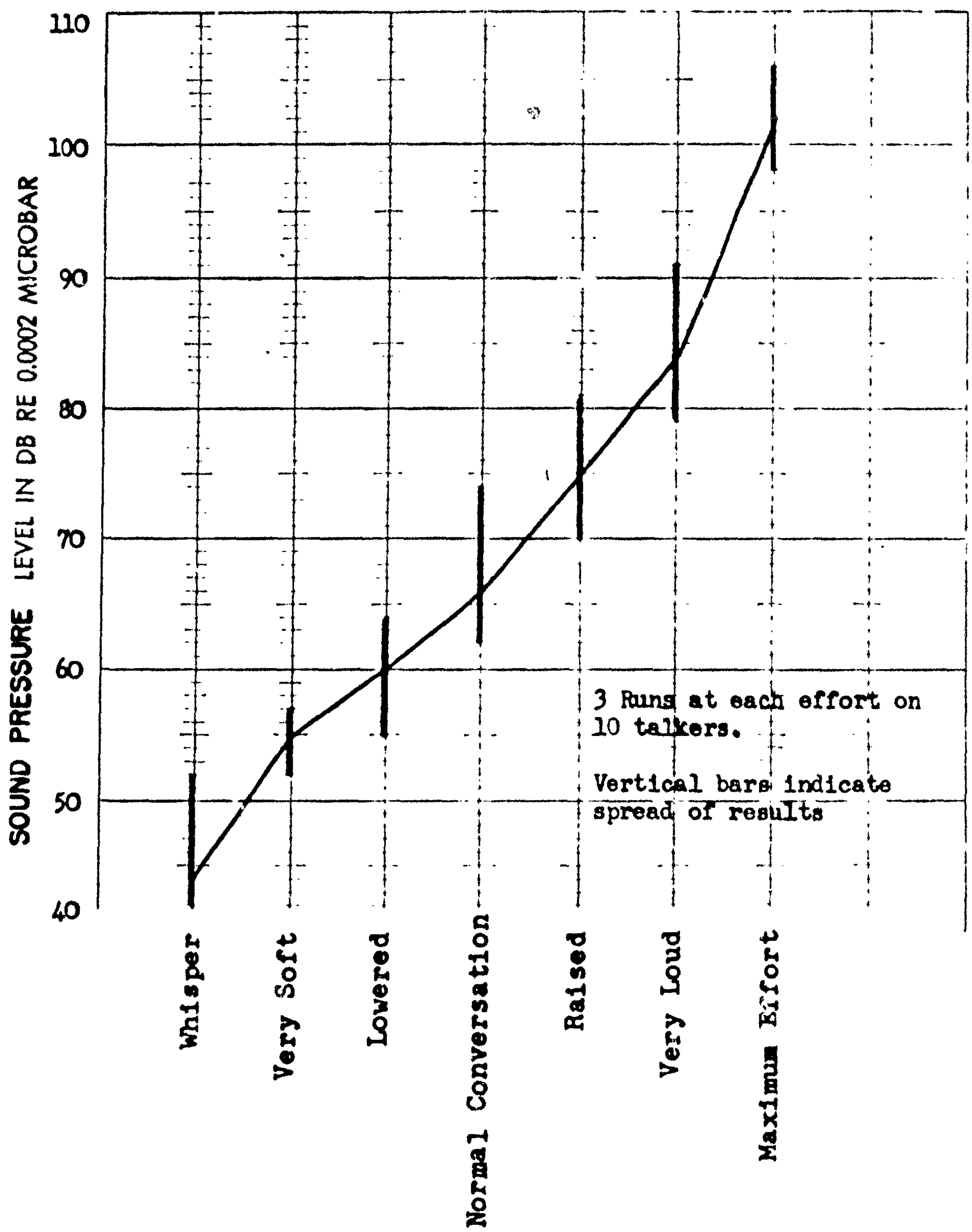
Subject: TW

Microphone 1/8" from lips.

Figure A4-8

SOUND PRESSURE LEVEL 1 FOOT FROM THE LIPS FOR VARIOUS

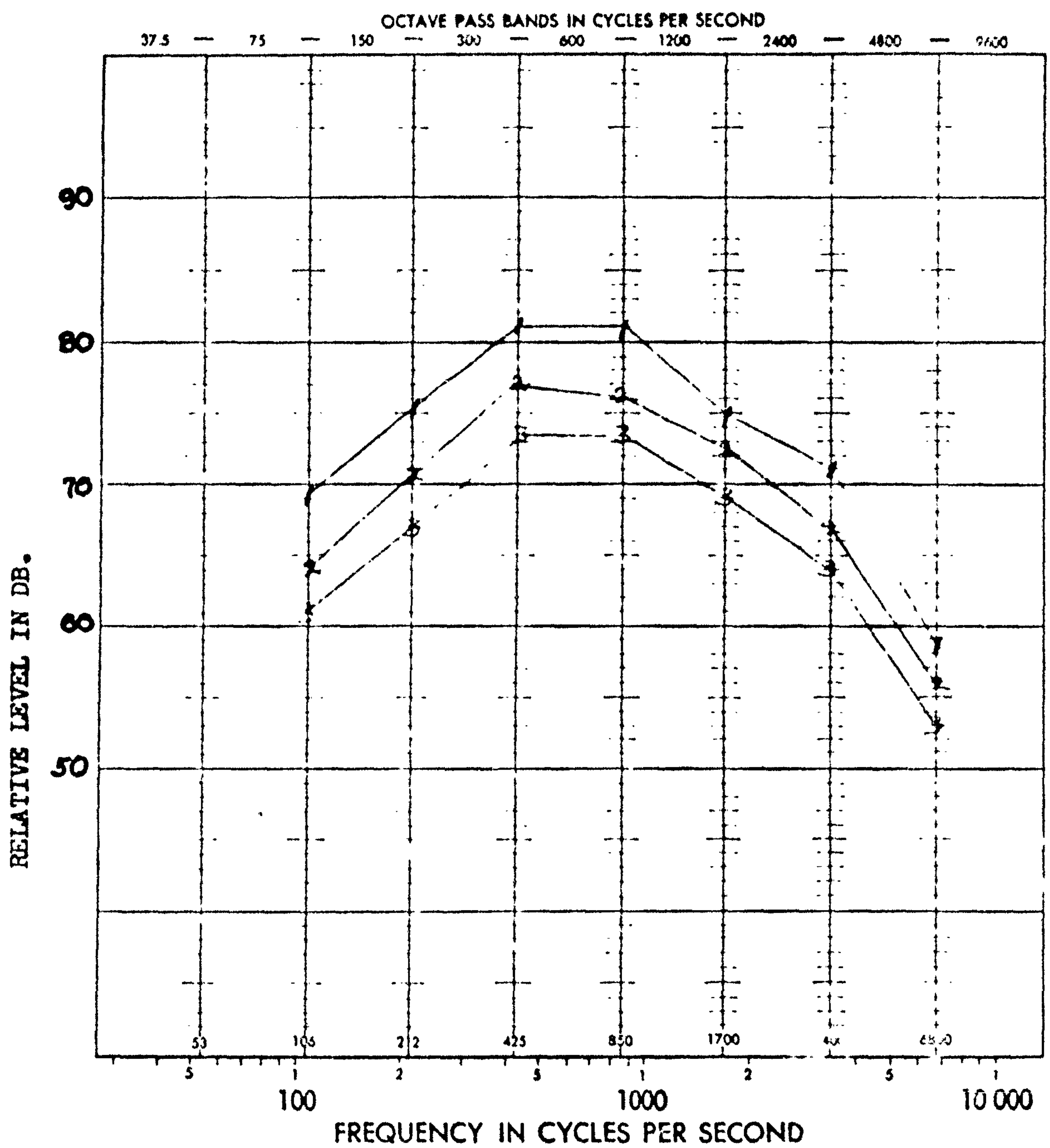
SUBJECTIVE SPEECH LEVELS



SUBJECTIVE SPEECH LEVEL

Figure A4-9

SOUND PRESSURE LEVEL AS FUNCTION OF DISTANCE FROM LIPS FOR A CONSTANT SPEAKING LEVEL.



Curve 1: Microphone at lips.

Curve 2: Microphone ½" from lips

Curve 3: Microphone 1" from lips

Note: 640 AA pressure-probe microphone used.

Figure A4-10

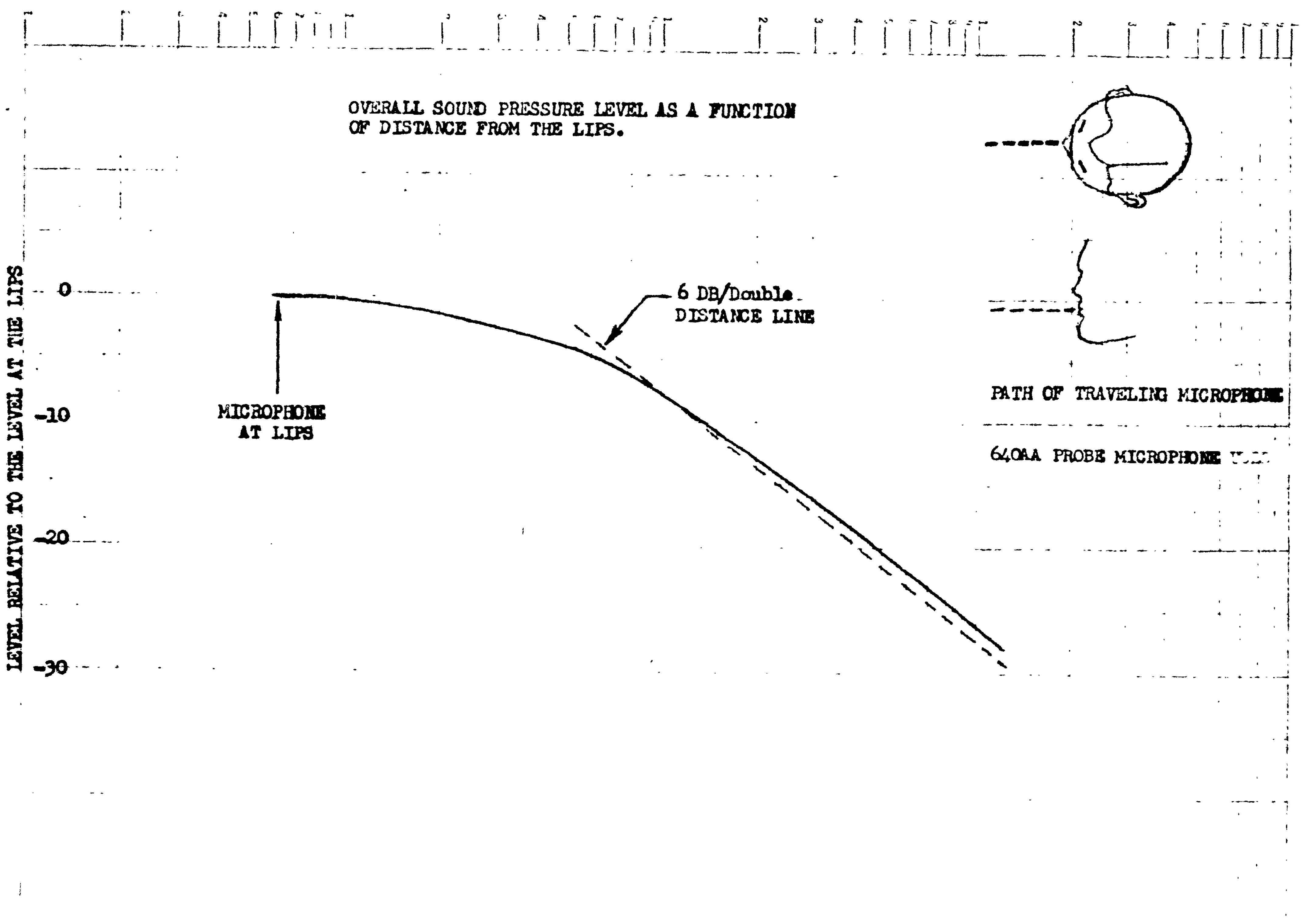
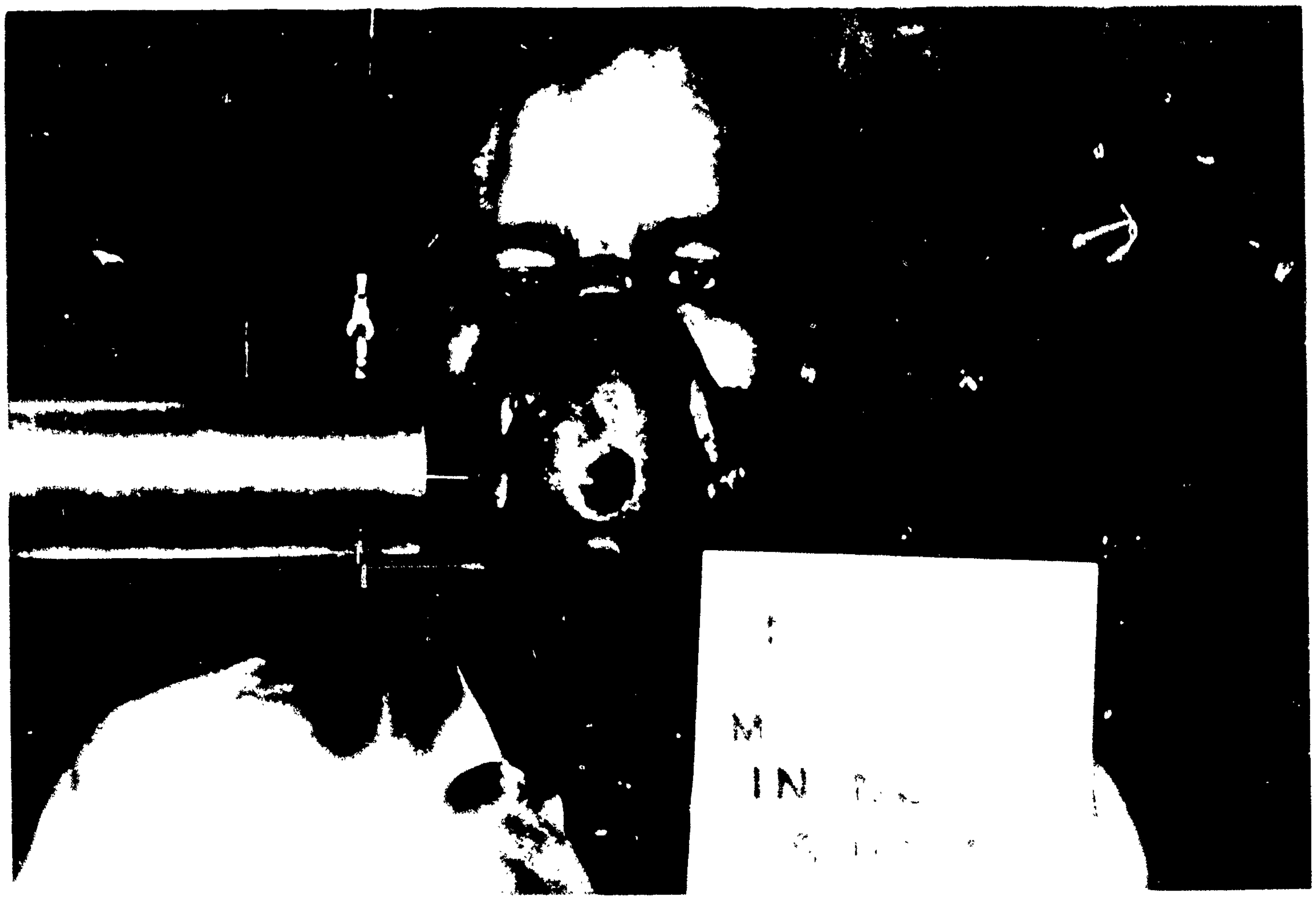
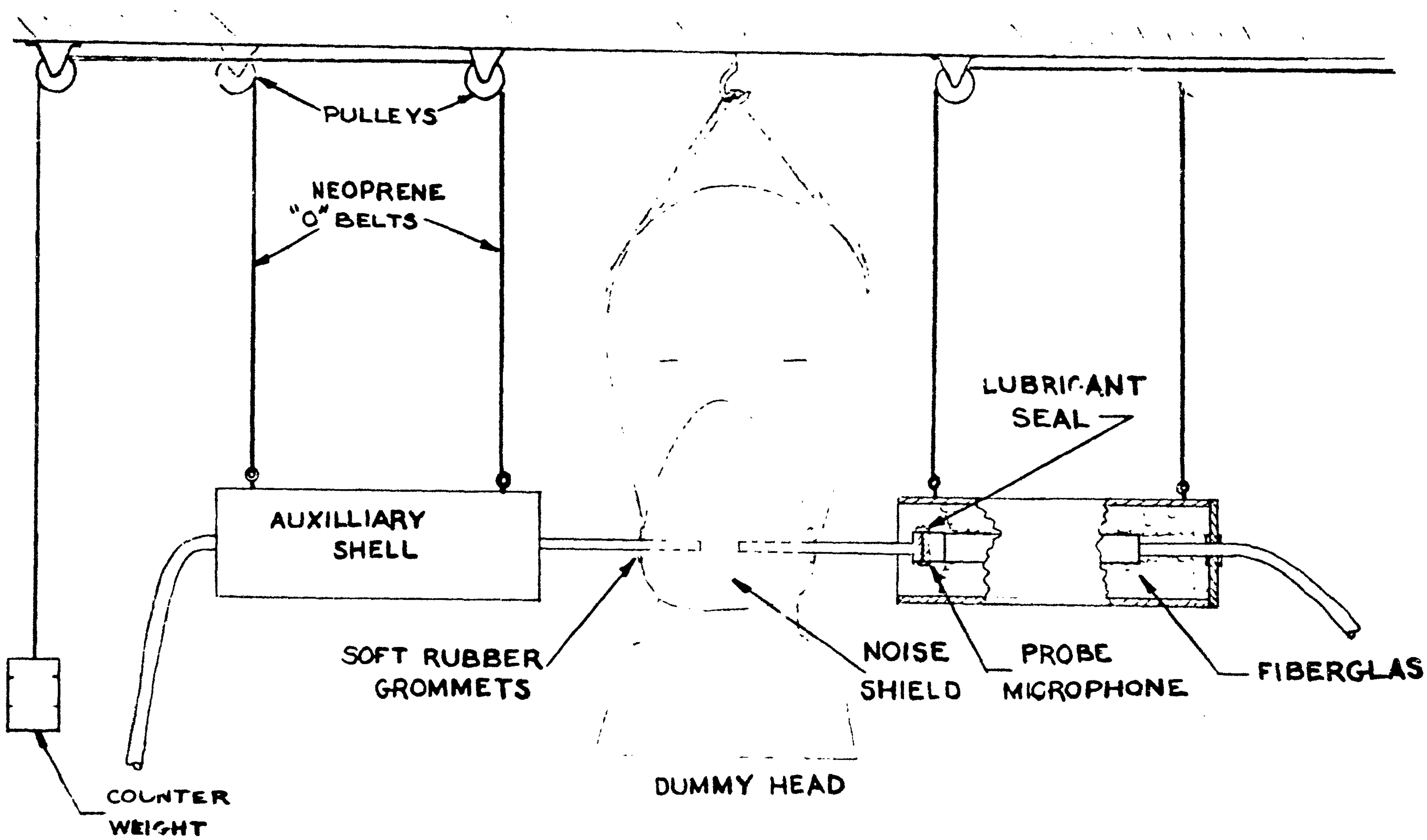

OVERALL SOUND PRESSURE LEVEL AS A FUNCTION
OF DISTANCE FROM THE LIPS.
6 DB/Double
DISTANCE LINE
MICROPHONE
AT LIPS
PATH OF TRAVELING MICROPHONE
640AA PROBE MICROPHONE
0
-10
-20
-30
LEVEL RELATIVE TO THE LEVEL AT THE LIPS

Figure A4-11

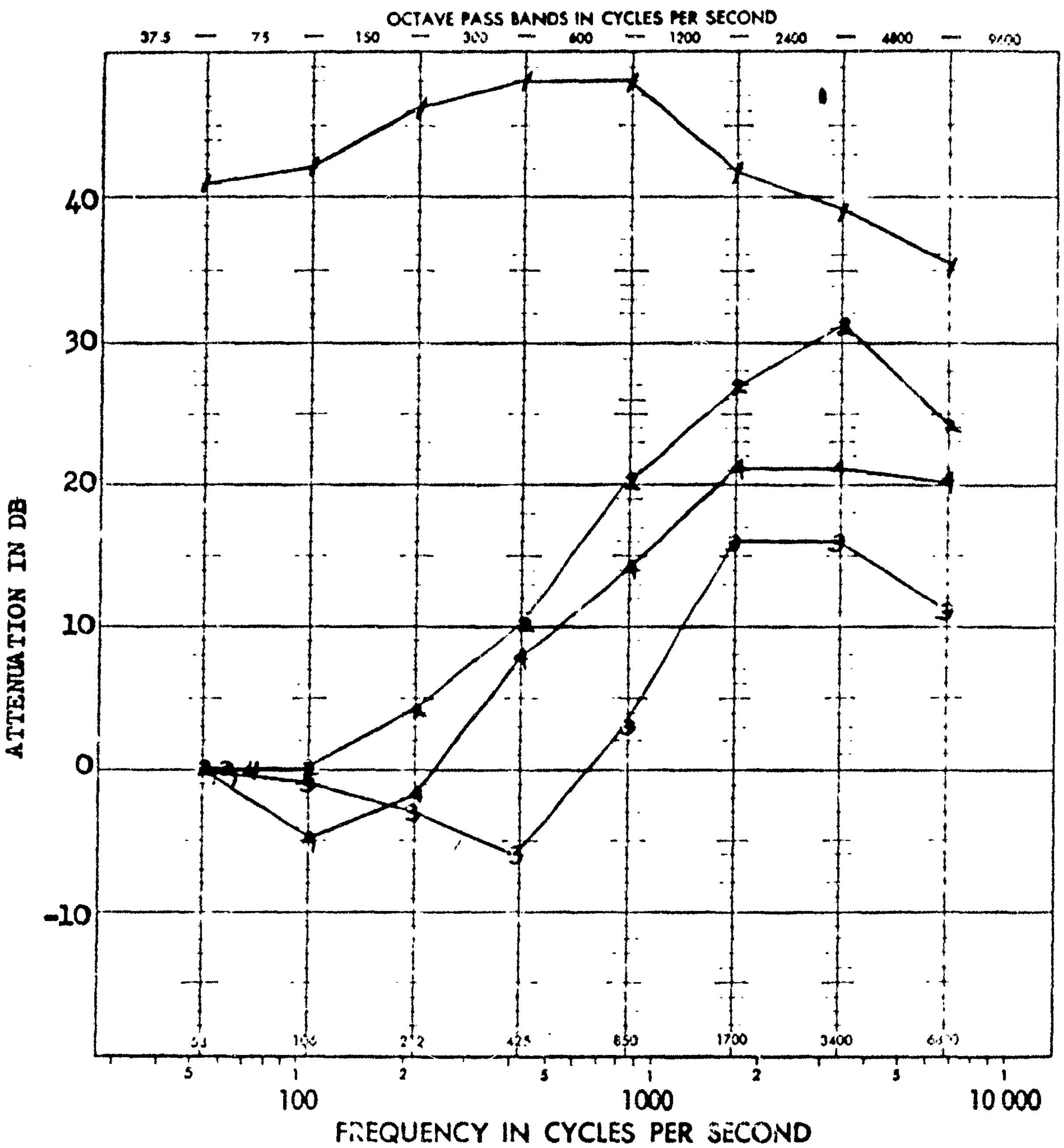




APPARATUS USED IN PHYSICAL MEASUREMENTS

Figure A4-13

ATTENUATION OF WEAL FIBERGLAS NOISE SHIELD



Curve 1: Noise shield completely sealed to dummy head.

Curve 2: Noise shield sealed as well as possible on human subject. No leak. Subject held breath during test.

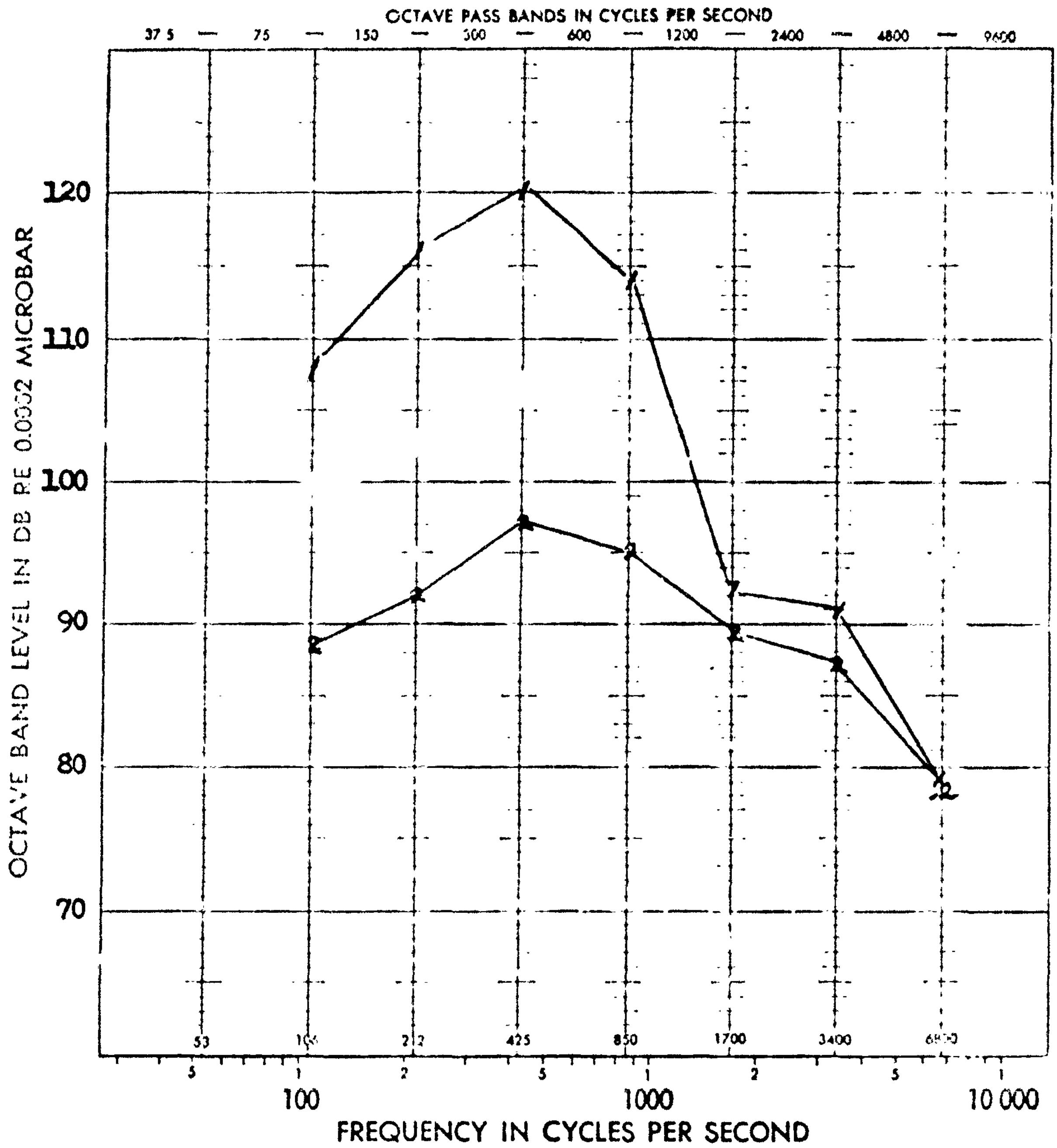
Curve 3: Noise shield with 1/4" wide by 3" long leak at subject's chin.

Curve 4: Noise shield with 3" long 3/8" I.D. tube protruding from bottom of shield providing leak.

Note: Data presented is averaged for 3 subjects. Leak is necessary for subject's breathing.

Figure A4-14

LONG TIME AVERAGE SPEECH SPECTRUM USING PRESSURE MICROPHONE IN OPEN AND IN WEAL FIBERGLAS NOISE SHIELD

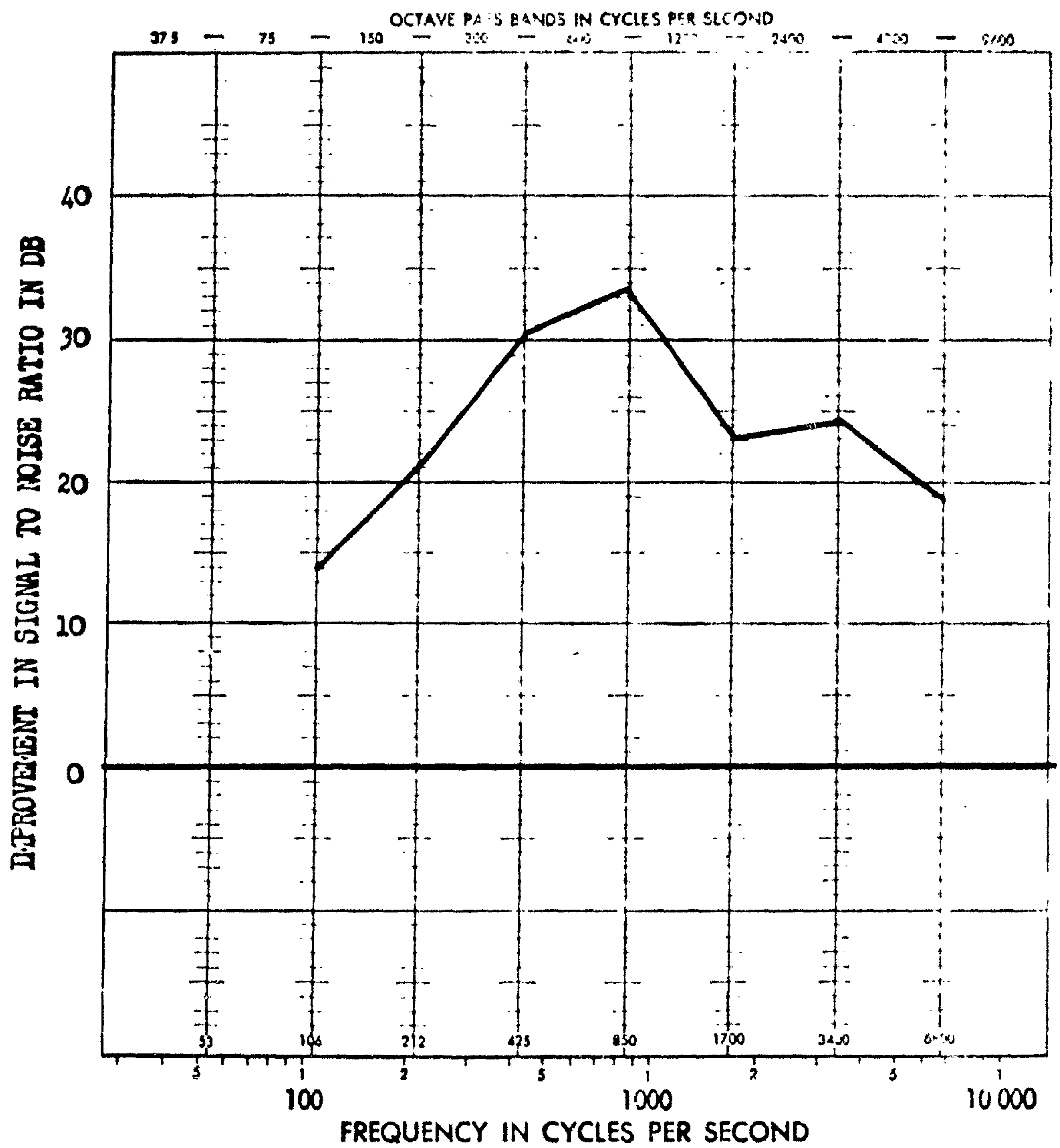


Curve 1: Pressure microphone in noise shield
Curve 2: Pressure microphone in open at the lips

Note: This is typical for all subjects.

Figure A4-15

IMPROVEMENT IN SIGNAL TO NOISE RATIO OF WEAL PRESSURE MICROPHONE IN FIBERGLAS NOISE SHIELD RELATIVE TO THE REFERENCE SYSTEM. SPEAKER: TW.

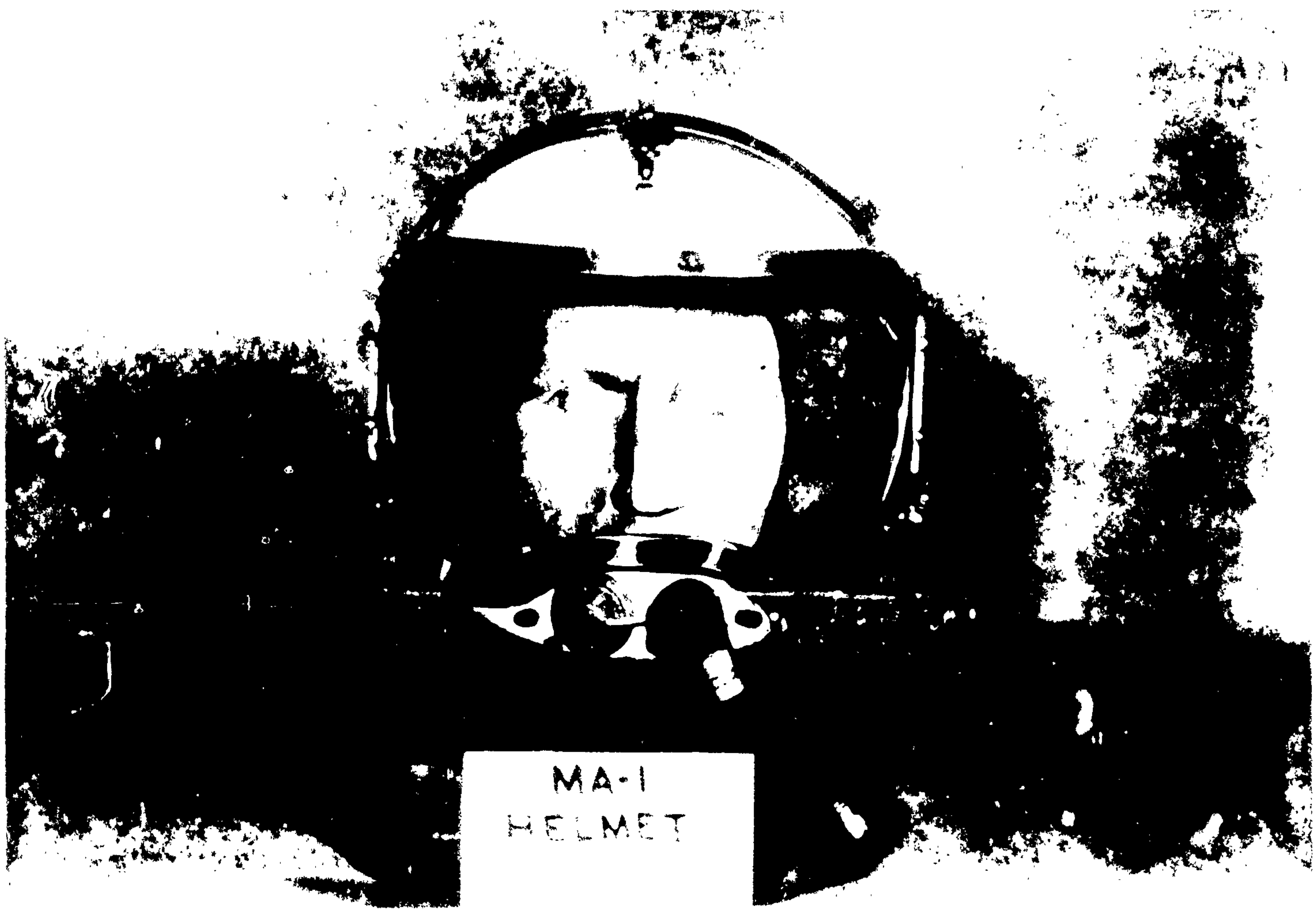


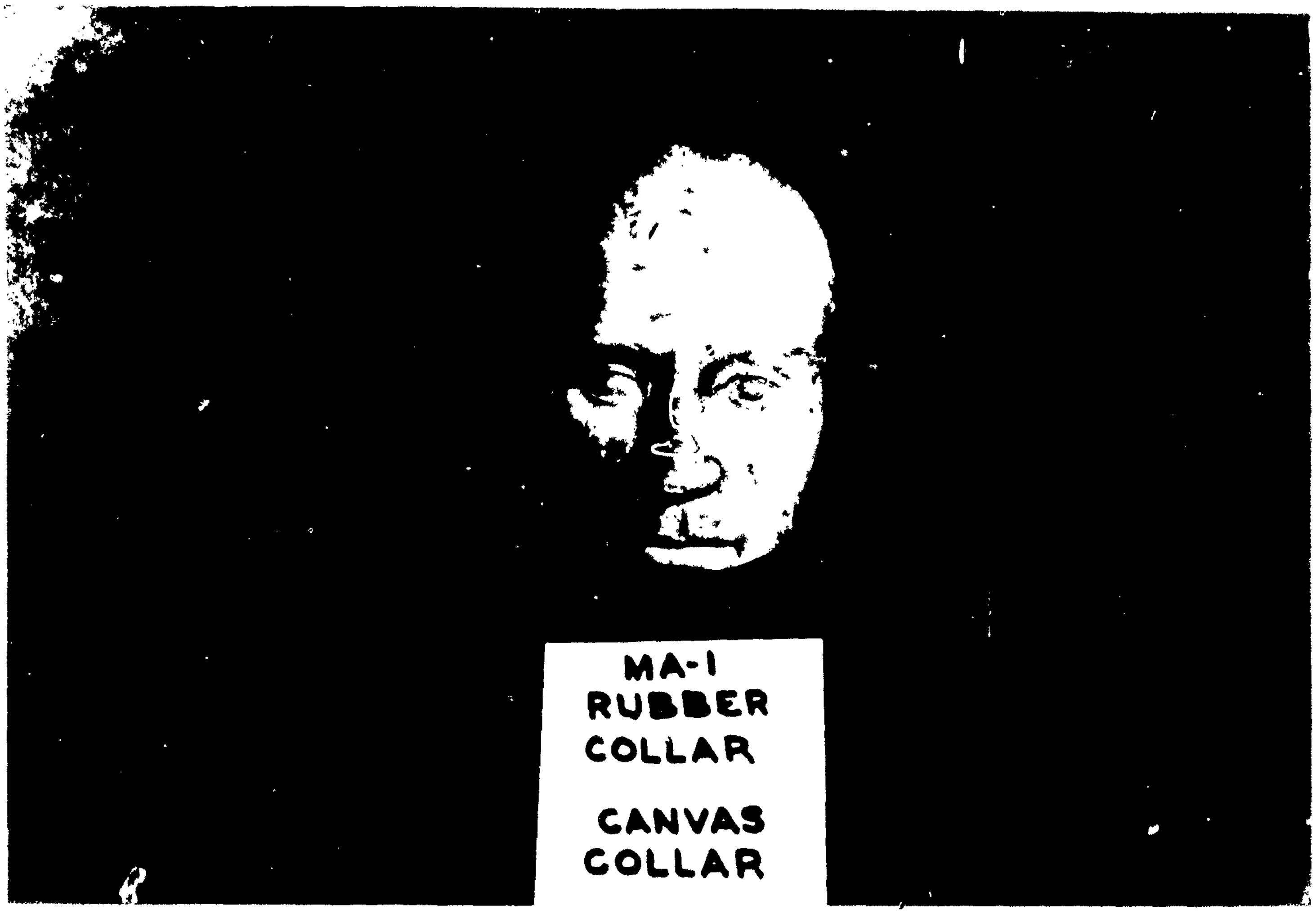
WEAL probe tube pressure microphone used.

Note: Leak provided by 3" long 3/8" I.D. tube.

Reference system is a Western Electric 640AA pressure probe microphone at the lips in the open.

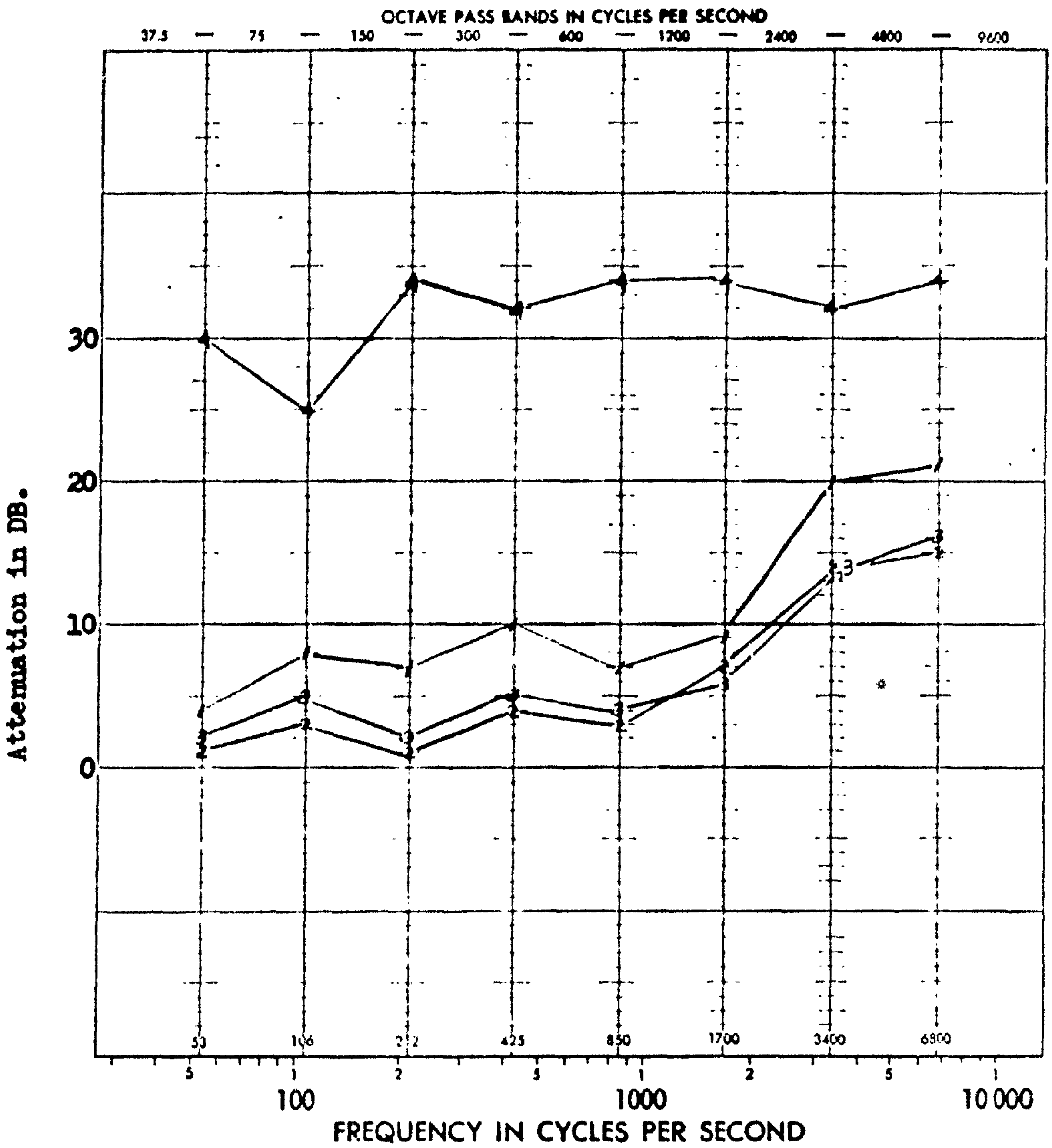
Figure A4-16


MA-1
HELMET


MA-1
RUBBER
COLLAR
CANVAS
COLLAR


640AA
PRESSURE
PROBE
MICROPHONE
IN MA-1
HELMET

ATTENUATION OF MA-1 HELMET ON DUMMY HEAD.

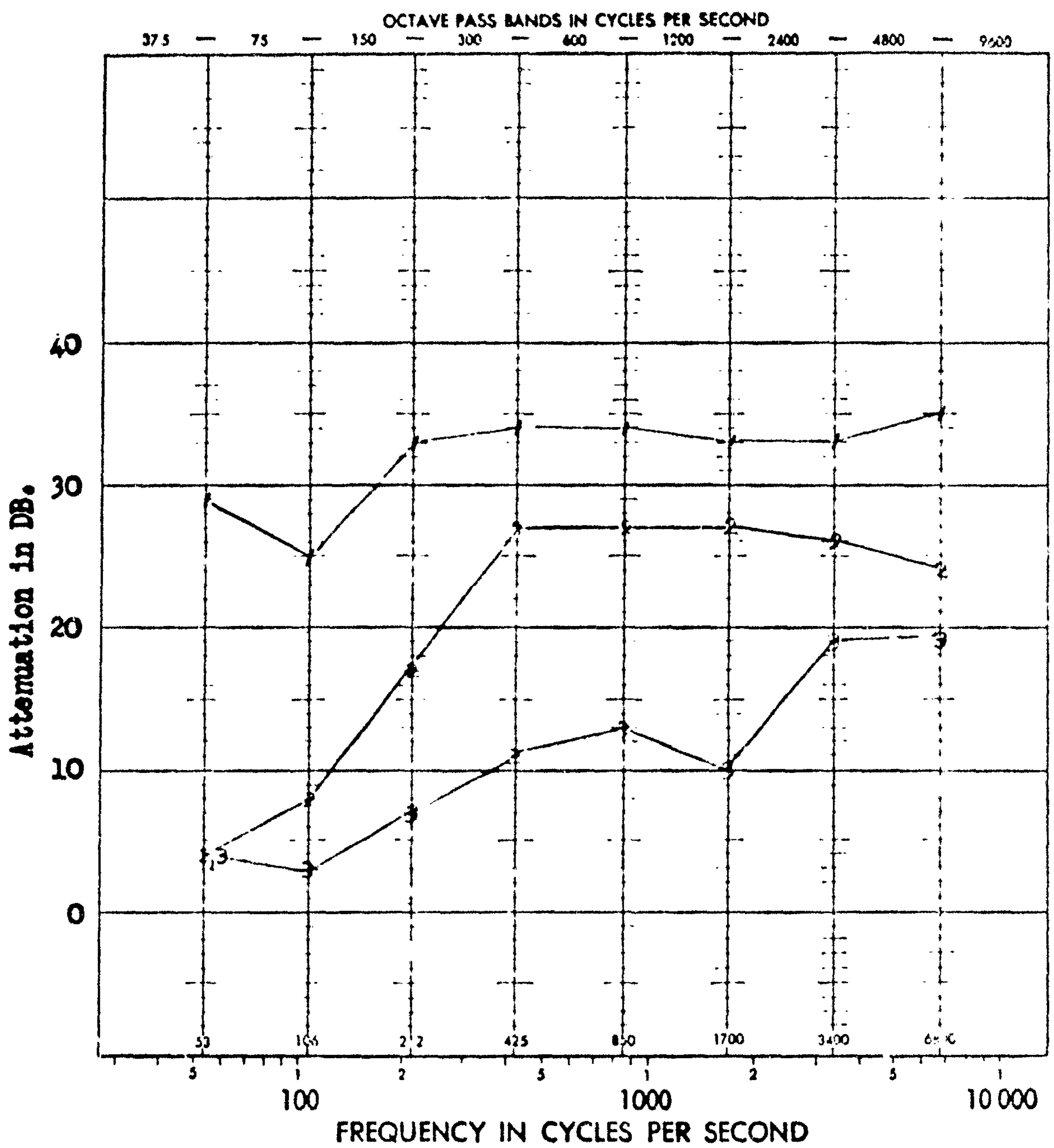


Curve 1: Attenuation with canvas and rubber collar attached to helmet.
Curve 2: Attenuation with rubber collar only.
Curve 3: Attenuation with canvas collar only.
Curve 4: Attenuation of helmet sealed to steel plate.

Note: Attenuation measured at lip position

Figure A4-20

EFFECT OF EXHAUST VALVE ON ATTENUATION OF MA-1 HELMET.



Curve 1: Attenuation of MA-1 helmet when clamped to steel plate. Exhaust valve closed.

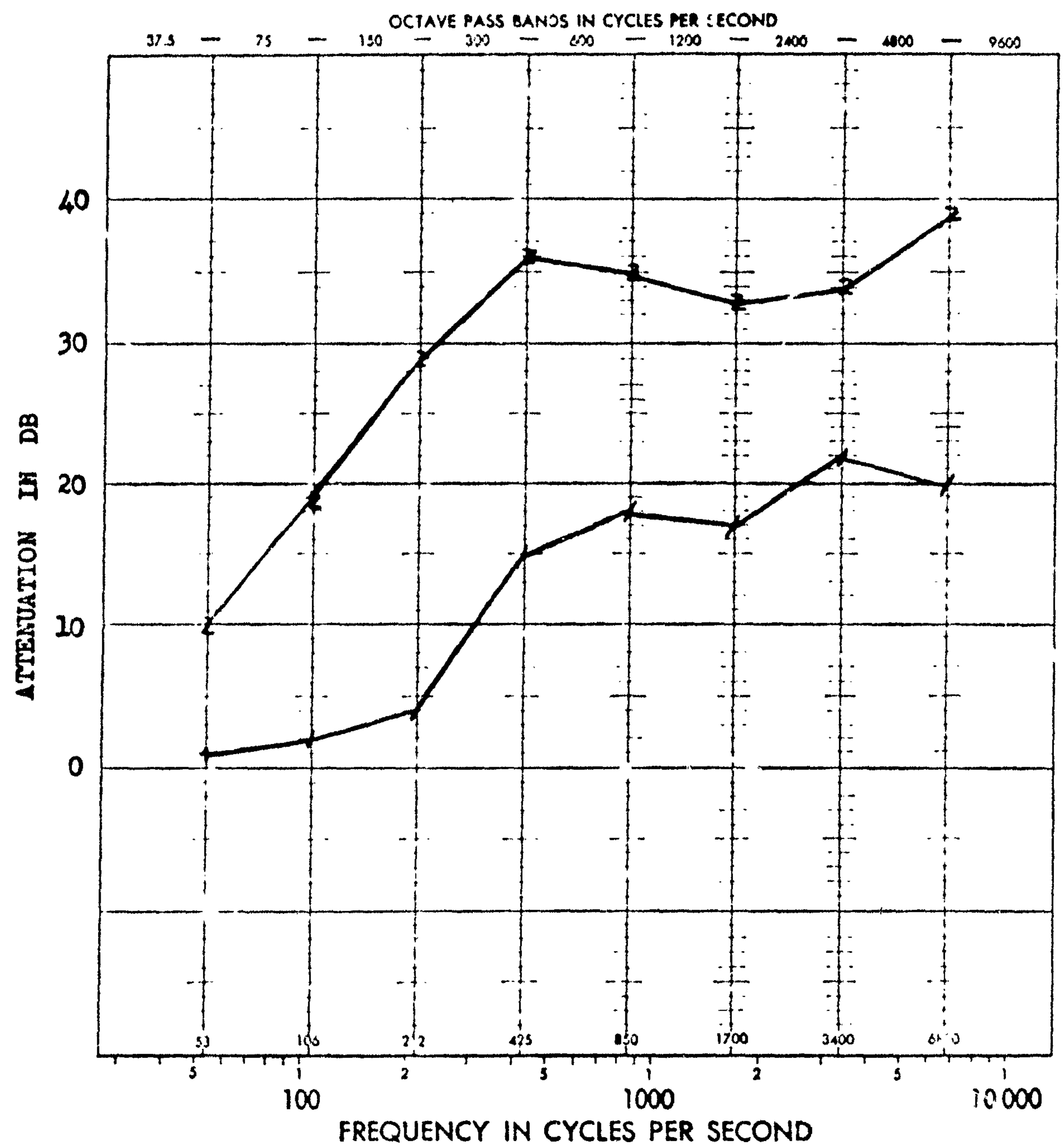
Curve 2: Attenuation of MA-1 helmet when clamped to steel plate. Exhaust valve opened slightly.

Curve 3: Attenuation of MA-1 helmet on a human subject. Average of 6 runs on 6 observers.

Note: Attenuation measured at lip position

Figure A4-21

ATTENUATION OF MA-3 HELMET



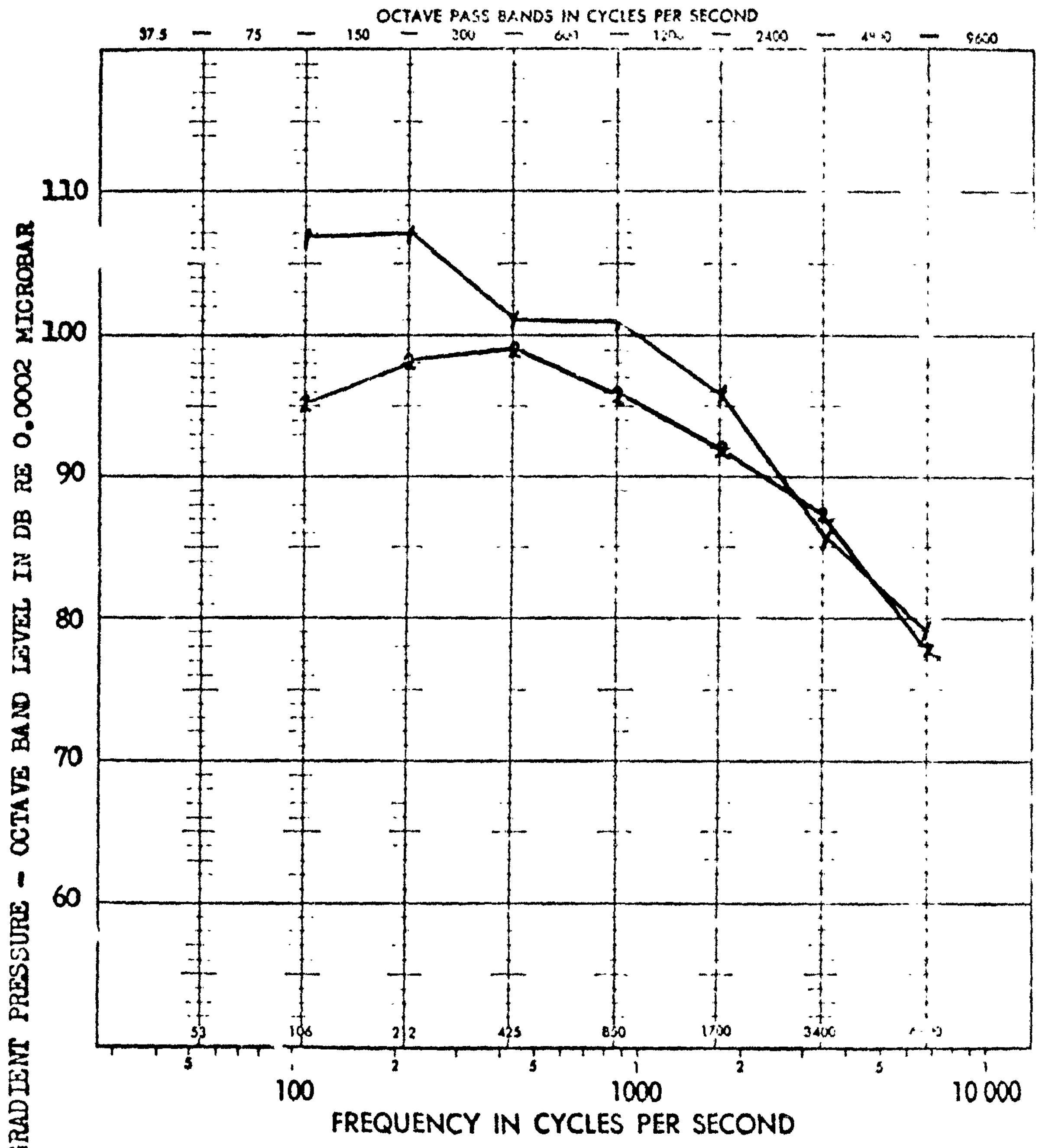
Curve 1: Attenuation of MA-3 Helmet. Average of four runs on two observers.

Curve 2: Attenuation of MA-3 Helmet clamped to rigid plate.

Note: Measured at lip position

Figure A4-22

LONG TIME AVERAGE SPEECH SPECTRA IN MA-1 HELMET
USING WEAL PRESSURE MICROPHONE. SPEAKER: TW.

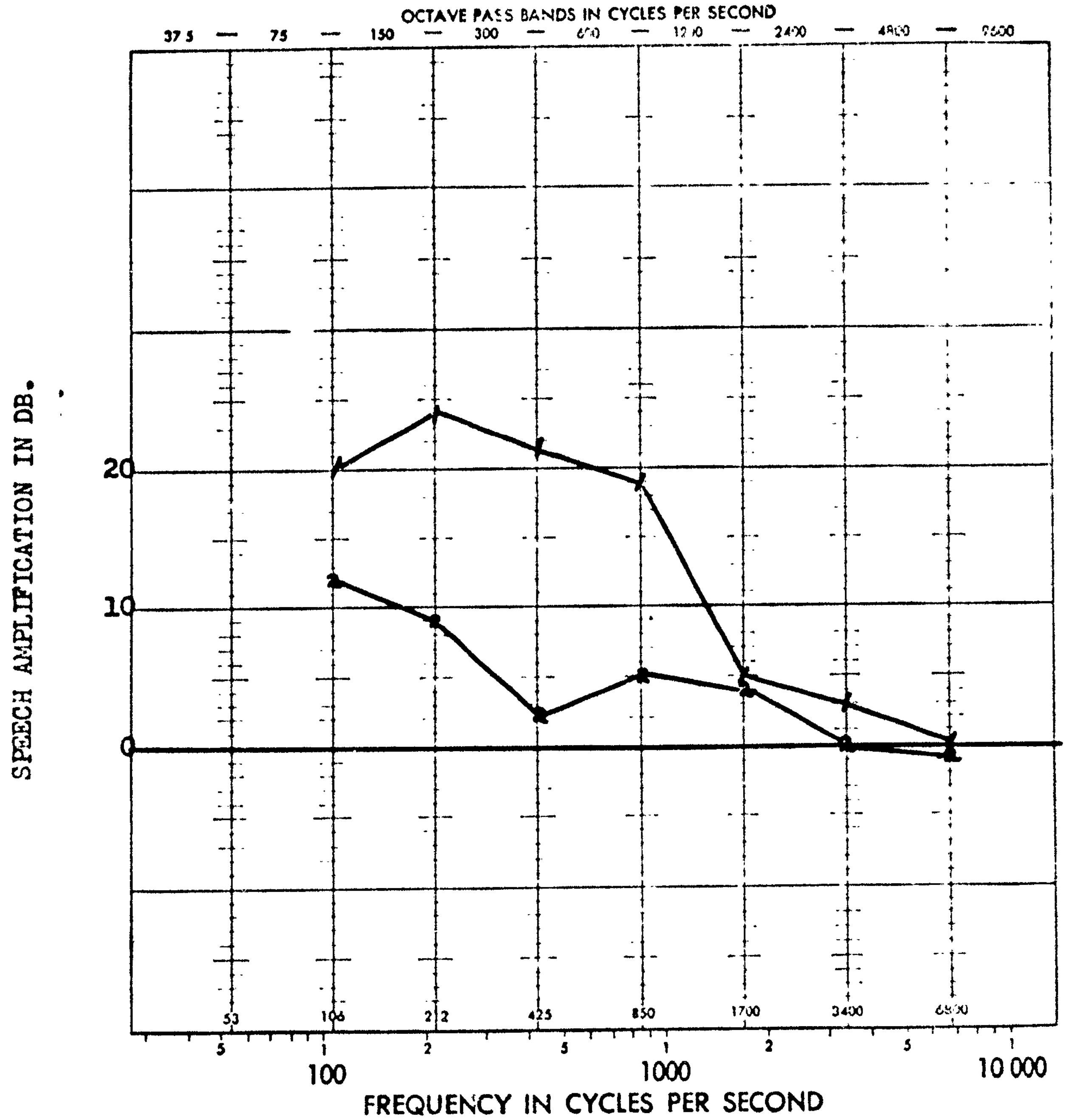


Curve 1: WEAL 640AA probe tube pressure microphone in MA-1 Helmet.

Curve 2: Same microphone in the "open." (Reference System)

Figure A4-23

COMPARISON OF SPEECH AMPLIFICATION IN NOISE SHIELD AND MA-1 HELMET



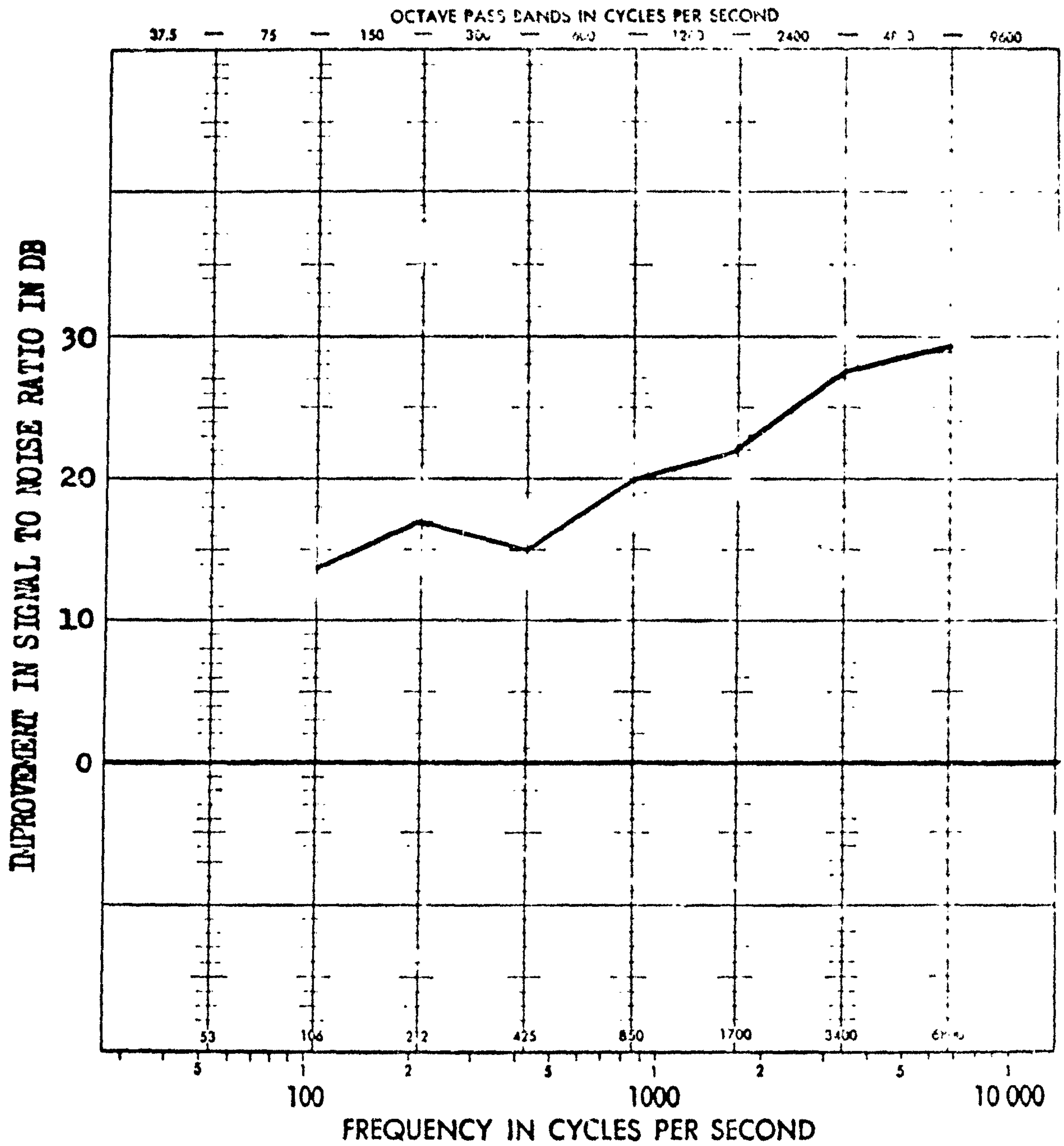
Curve 1: W.E. 640AA pressure probe mic in WEAL fiberglas noise shield

Curve 2: W.E. 640AA pressure probe mic in MA-1 helmet

Note: this is typical for all subjects.

Figure A4-24

IMPROVEMENT IN SIGNAL TO NOISE RATIO OF WEAL PRESSURE MICROPHONE IN MA-1 HELMET RELATIVE TO THE REFERENCE SYSTEM. SPEAKER: TW.

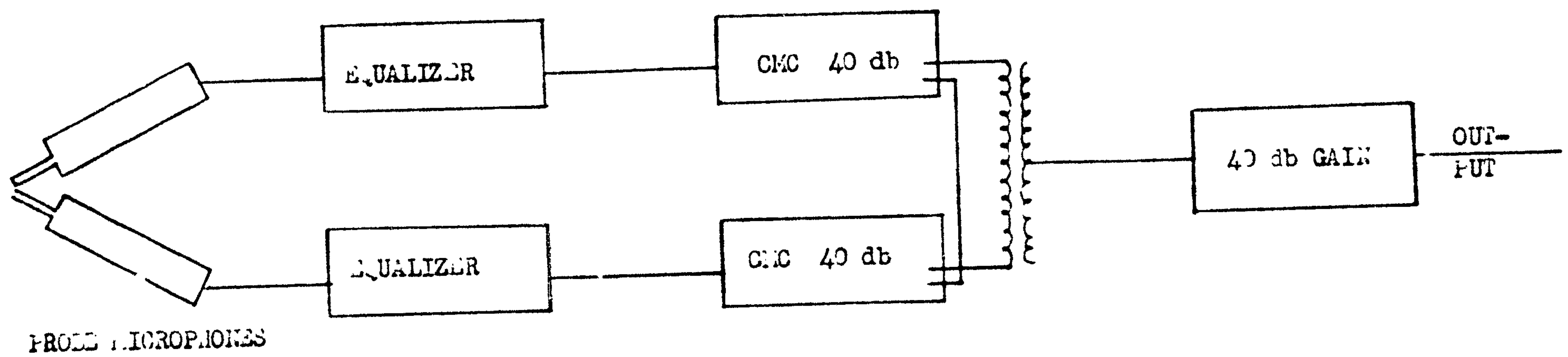


Note: WEAL 640AA probe tube pressure microphone used.

Reference system is a Western Electric 640AA pressure probe microphone at the lips in the open.

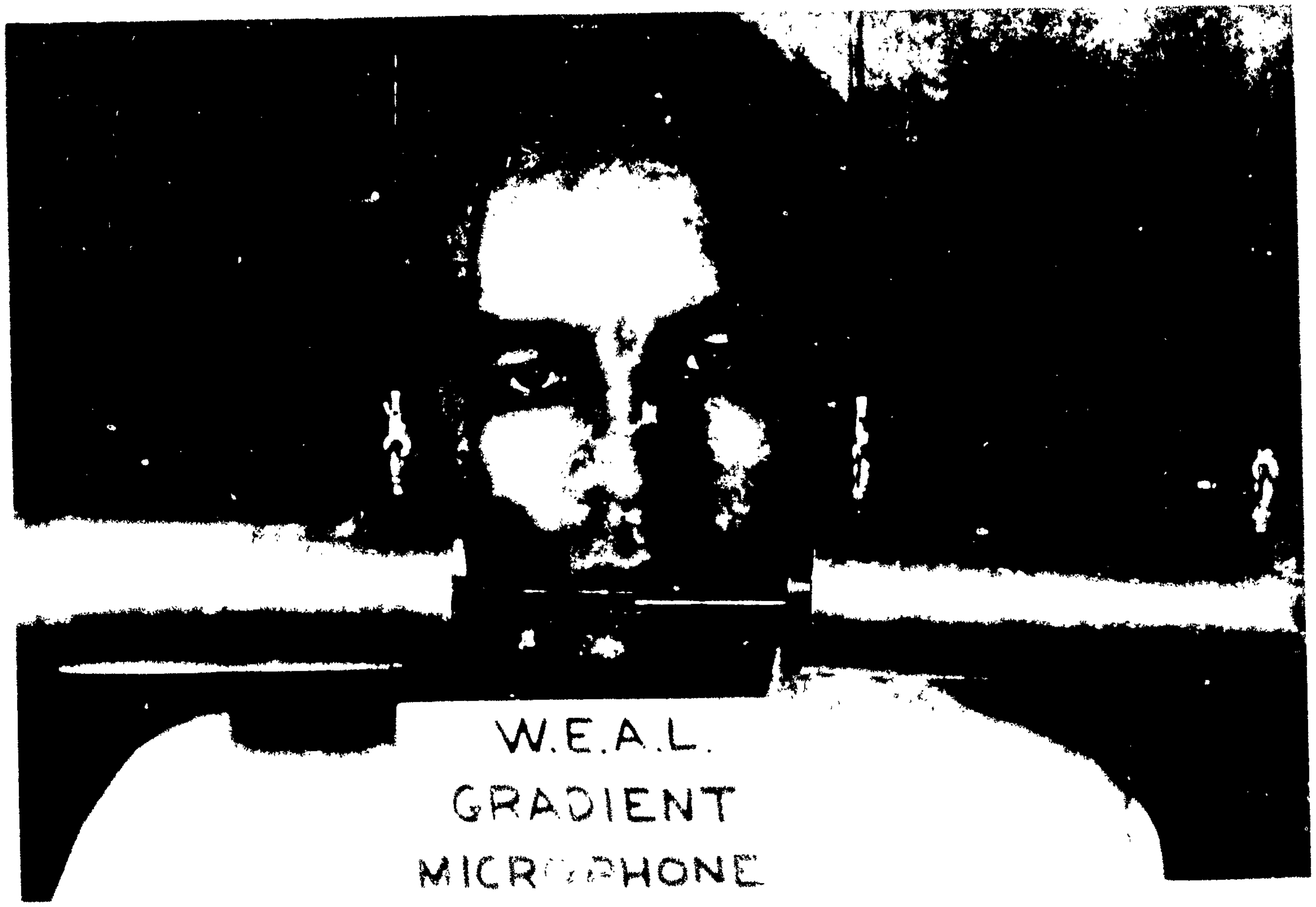
Note: This is typical for all subjects

Figure A4-25

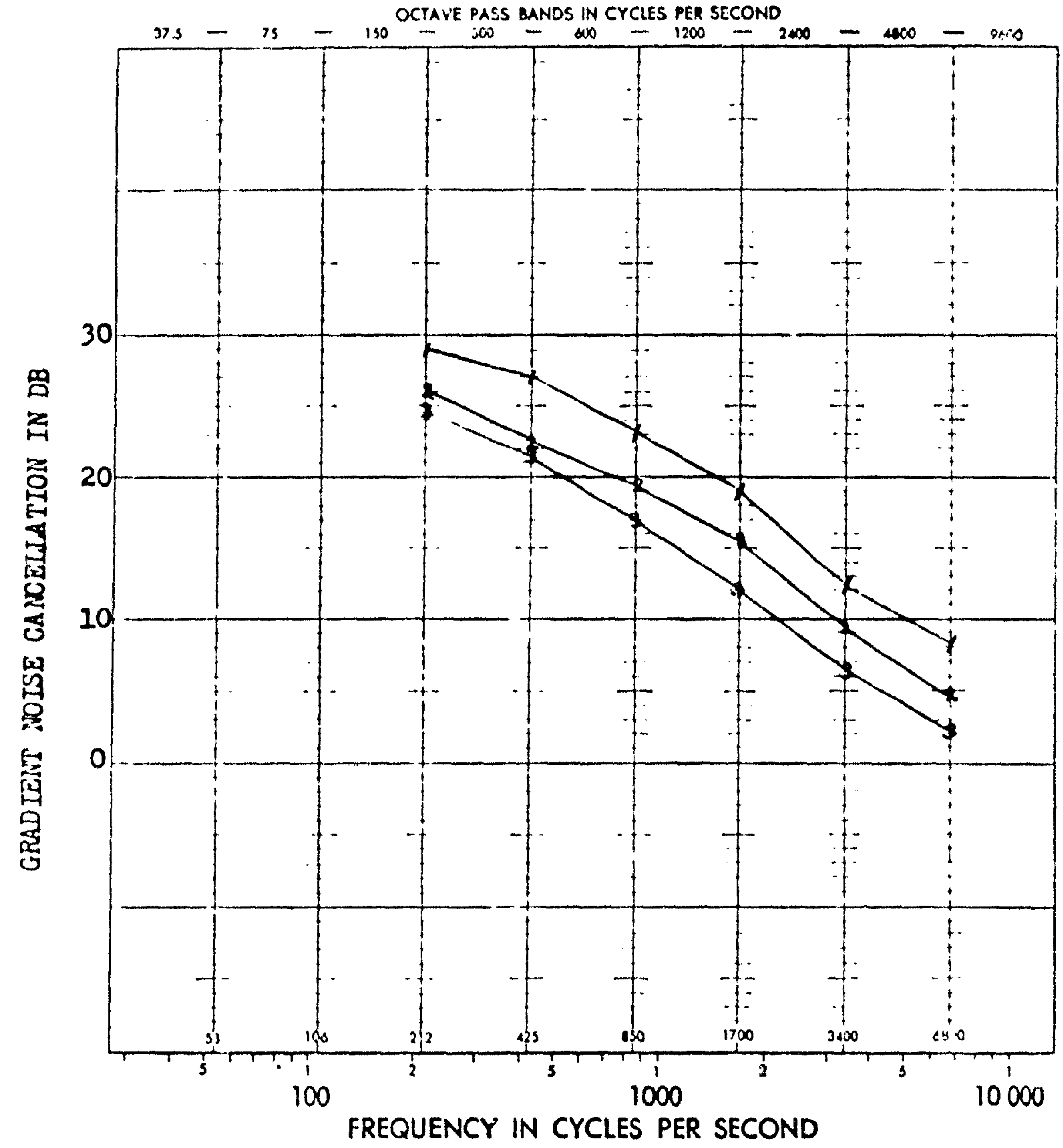


BLOCK DIAGRAM FOR GRADIENT MICROPHONE

Figure A4-26


W.E.A.L.
GRADIENT
MICROPHONE

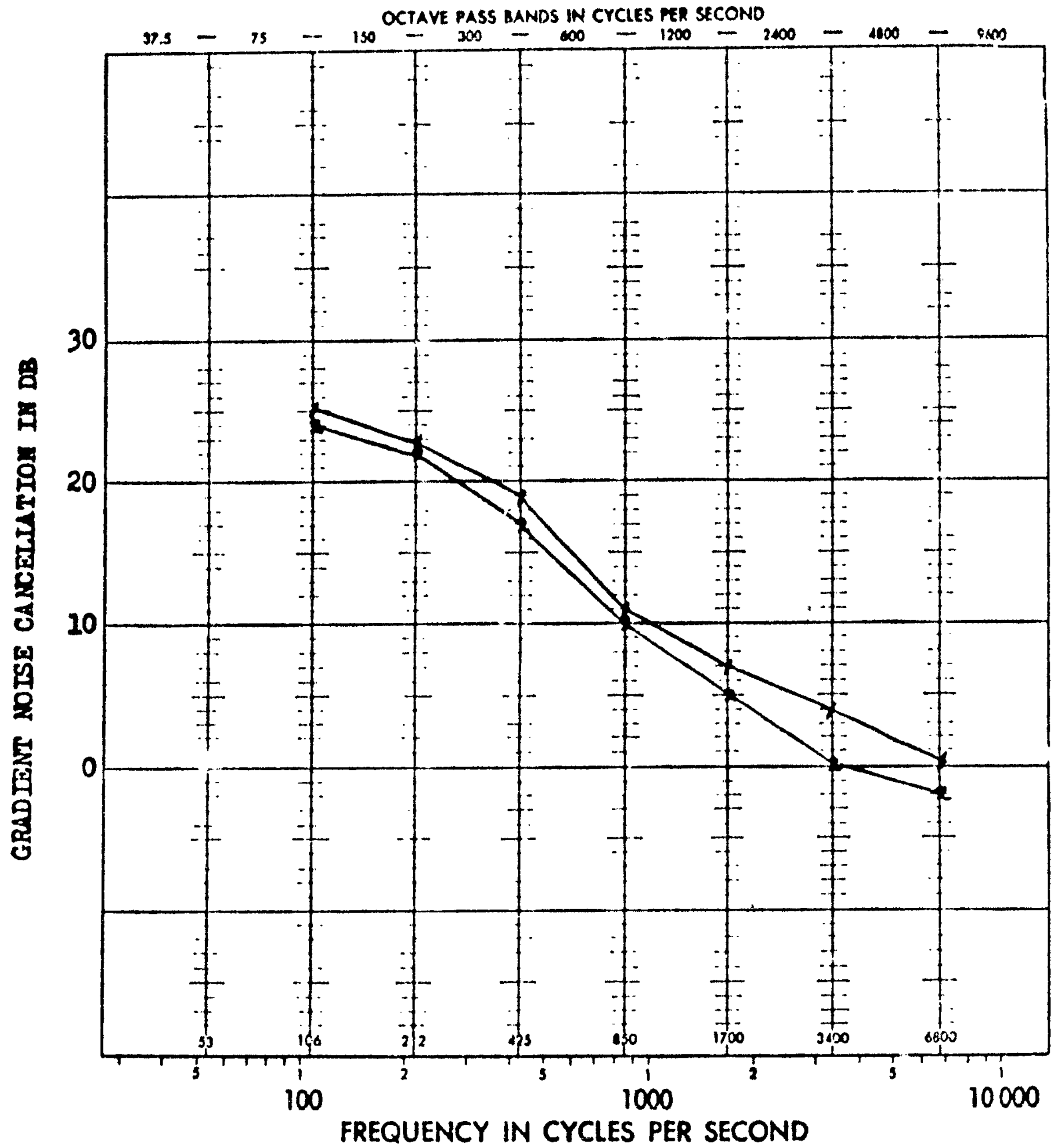
NOISE CANCELLATION OF 640AA PROBE GRADIENT
MICROPHONE AS A FUNCTION OF PROBE SPACING



Curve 1: Probe spacing - 3/16" (center to center).
Curve 2: Probe spacing - 5/16" (center to center).
Curve 3: Probe spacing - 1/2" (center to center).

Figure A4-28

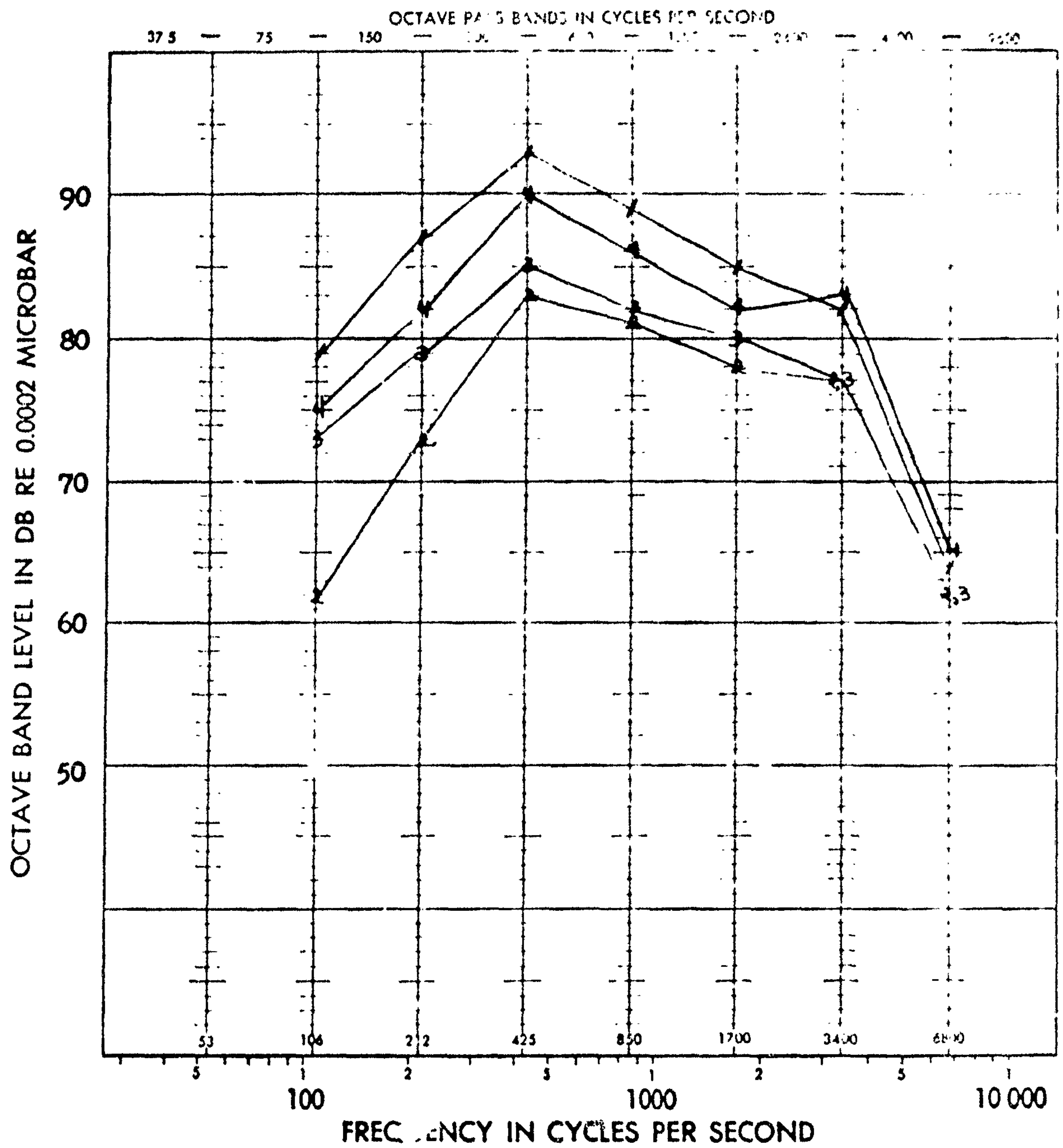
EFFECT OF BAFFLE ON NOISE CANCELLING PROPERTIES
OF A 640AA PROBE GRADIENT MICROPHONE



Curve 1: 3/8" probe spacing. No baffle.
Curve 2: 3/8" probe spacing. Baffle consisting of 25¢ piece suspended between probes.

Figure A4-29

EFFECT OF GRADIENT MICROPHONE PROBE SPACING ON LTA SPEECH SPECTRA. SPEAKER: RAB.

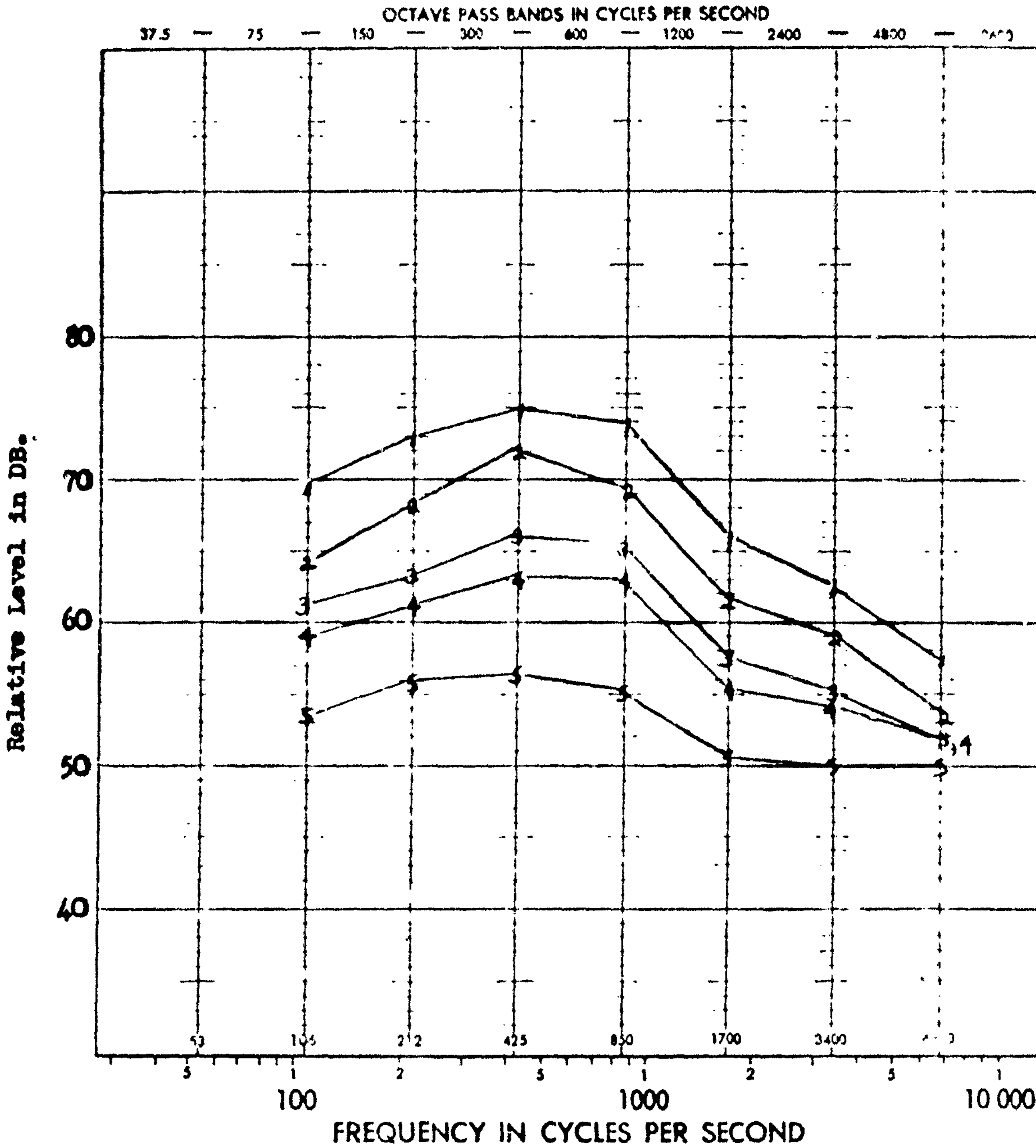


Curve 1: LTA speech spectra as measured by open pressure microphone at the lips (reference system).
Curve 2: LTA speech spectra. Gradient spacing– 3/16".
Curve 3: LTA speech spectra. Gradient spacing – 3/8".
Curve 4: LTA speech spectra. Gradient spacing – 3/4".

Note: Typical for all other subjects

Figure A4-30

RESPONSE OF WESTERN ELECTRO-ACOUSTIC LABORATORY GRADIENT MICROPHONE (1/4" SPACING) FOR CONSTANT SPEECH LEVEL VS DISTANCE FROM LIPS. ×

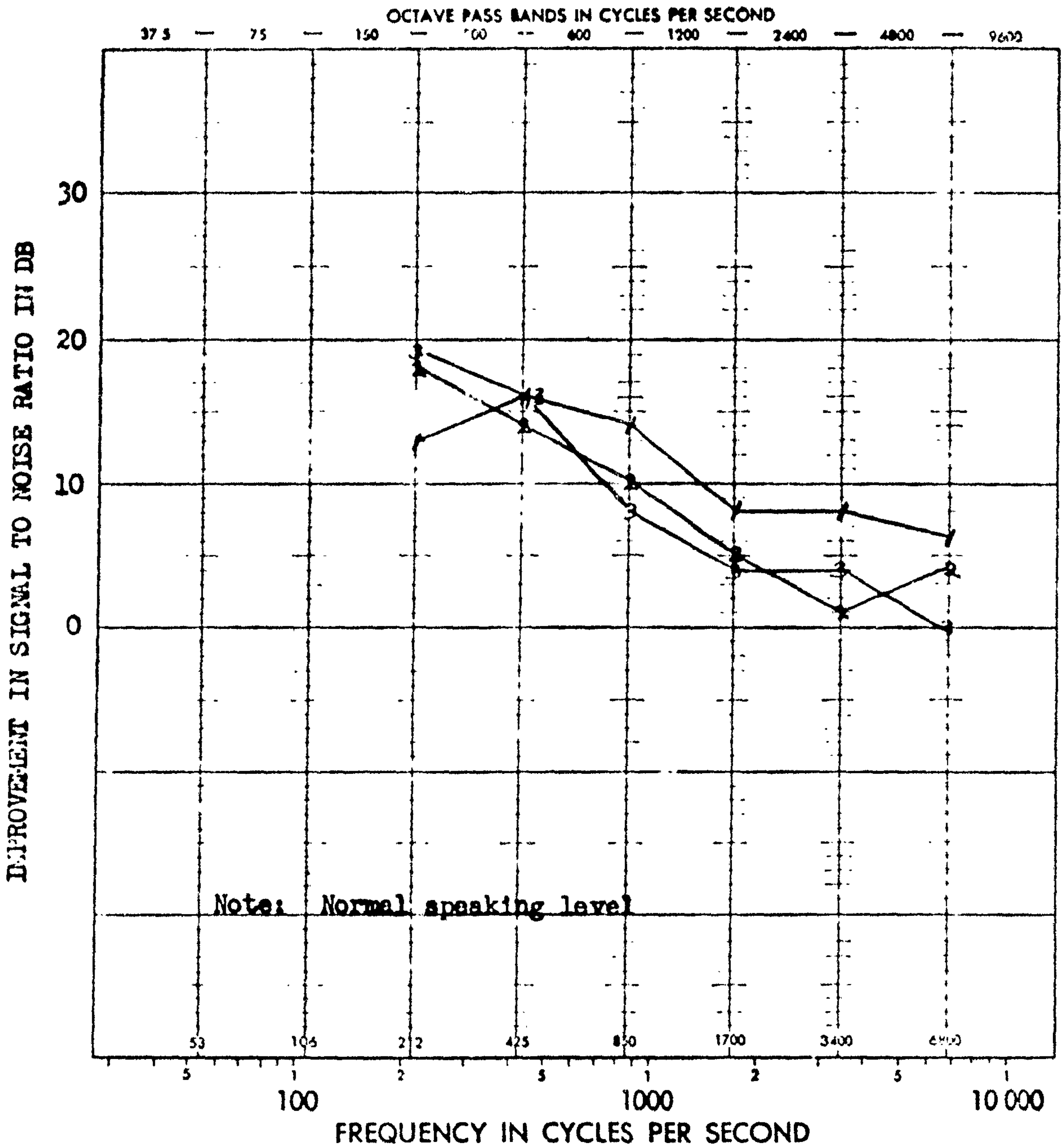


Curve 1: Microphone at lips.
Curve 2: Microphone 1/8" from lips.
Curve 3: Microphone 1/4" from lips.
Curve 4: Microphone 1/2" from lips.
Curve 5: Microphone 1" from lips.

Speaker: JPC. Typical for other subjects

Figure A4-31

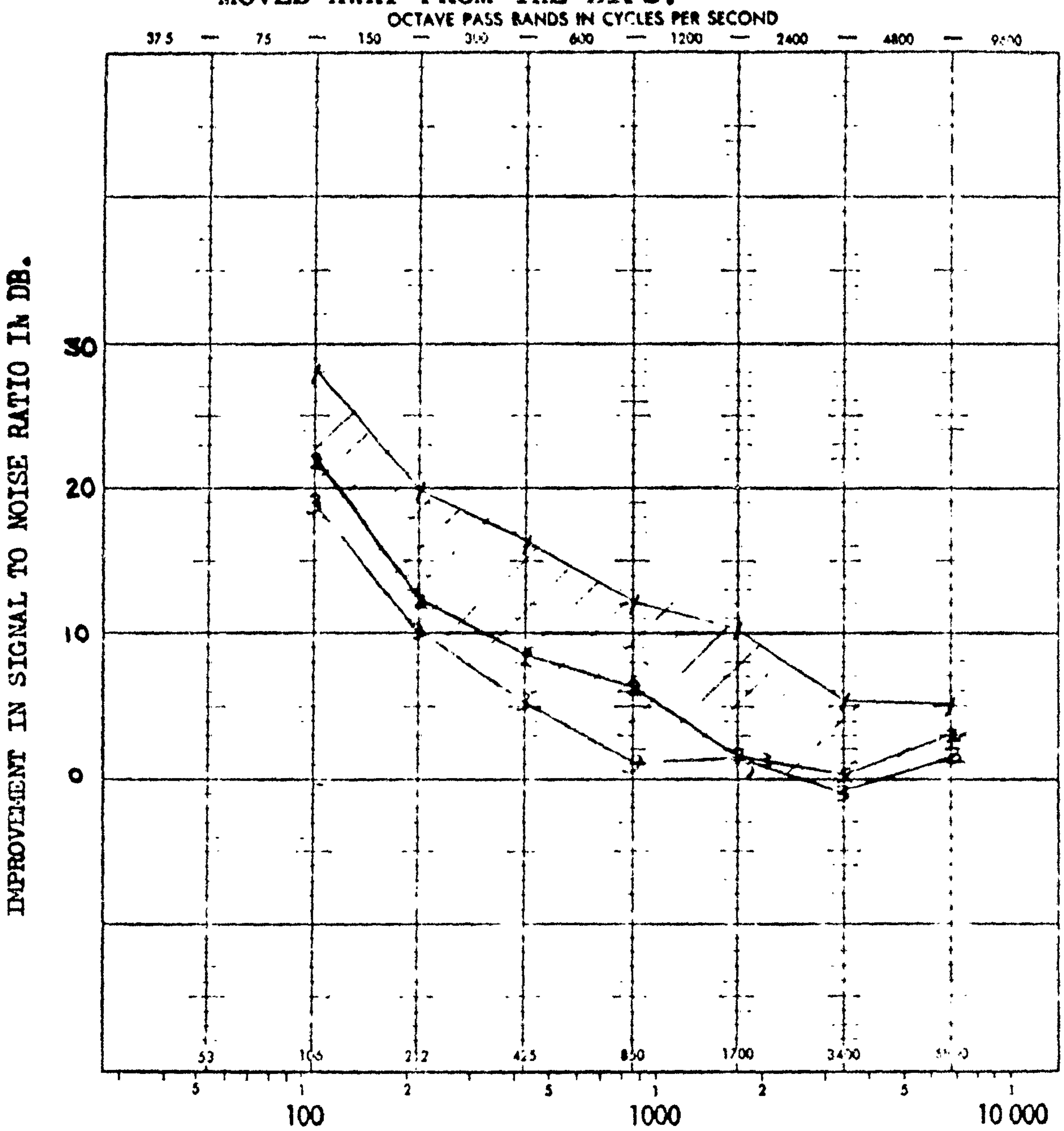
IMPROVEMENT IN SIGNAL TO NOISE RATIO OVER A PRESSURE MICROPHONE AT THE LIPS FOR PROBE GRADIENT MICROPHONE AS A FUNCTION OF PROBE SPACING



Curve 1: Probe spacing = 3/16"
Curve 2: Probe spacing = 3/8"
Curve 3: Probe spacing = 3/4"

Figure A4-32

EFFECT ON IMPROVEMENT IN SIGNAL TO NOISE RATIO OF A GRADIENT MICROPHONE RELATIVE TO AN OPEN PRESSURE MICROPHONE AS BOTH ARE MOVED AWAY FROM THE LIPS.



Curvel: Both microphones at lips

Curve 2: " " ½" from lips

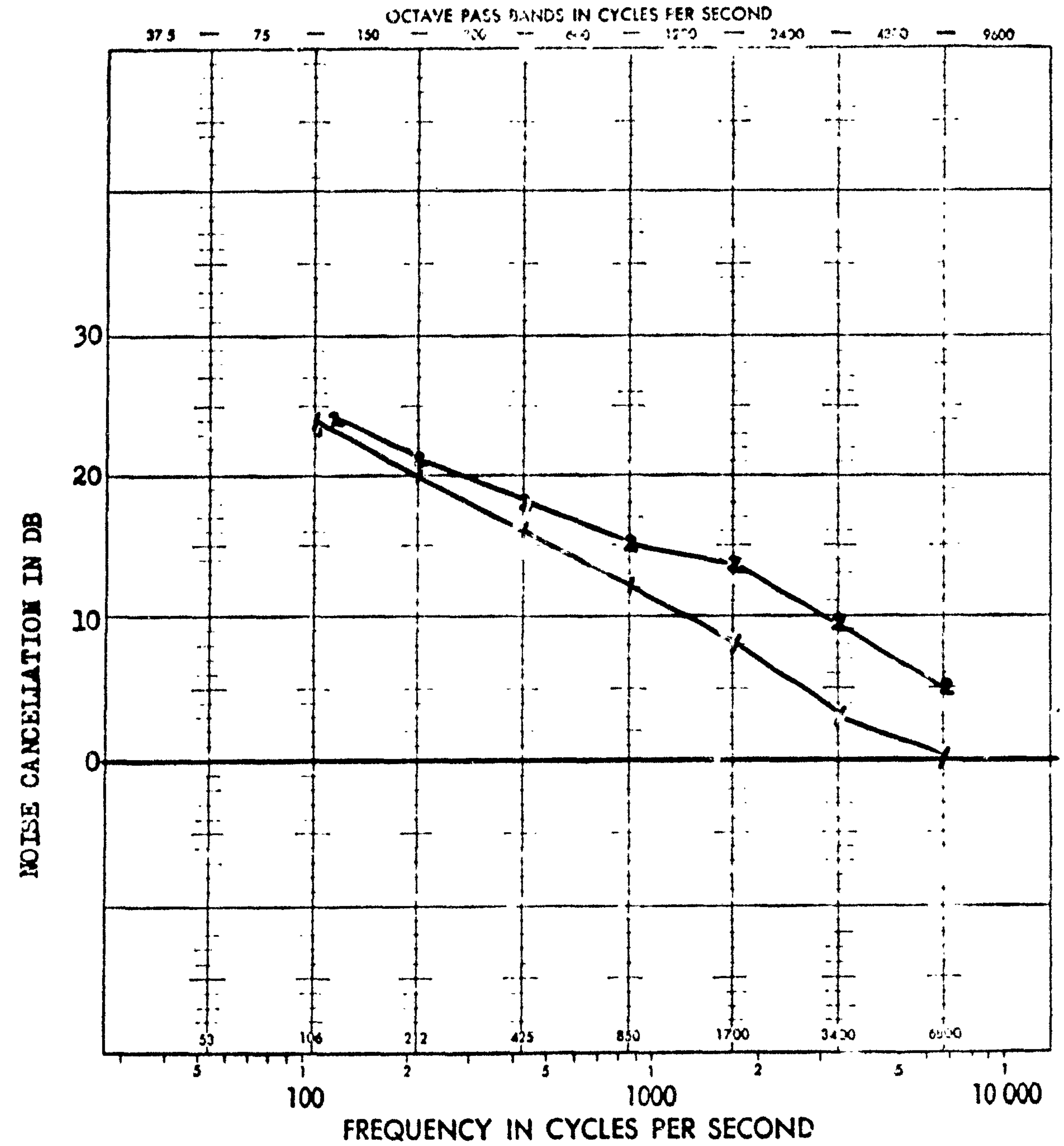
Curve 3: " " 1" from lips

Note: Relative to probe pressure microphone at same position

Figure A4-33

Figure 4.3

NOISE CANCELLATION OF GRADIENT MICROPHONE IN WEAL FIBERGLAS SHIELD AS COMPARED TO THE NOISE CANCELLATION IN THE OPEN



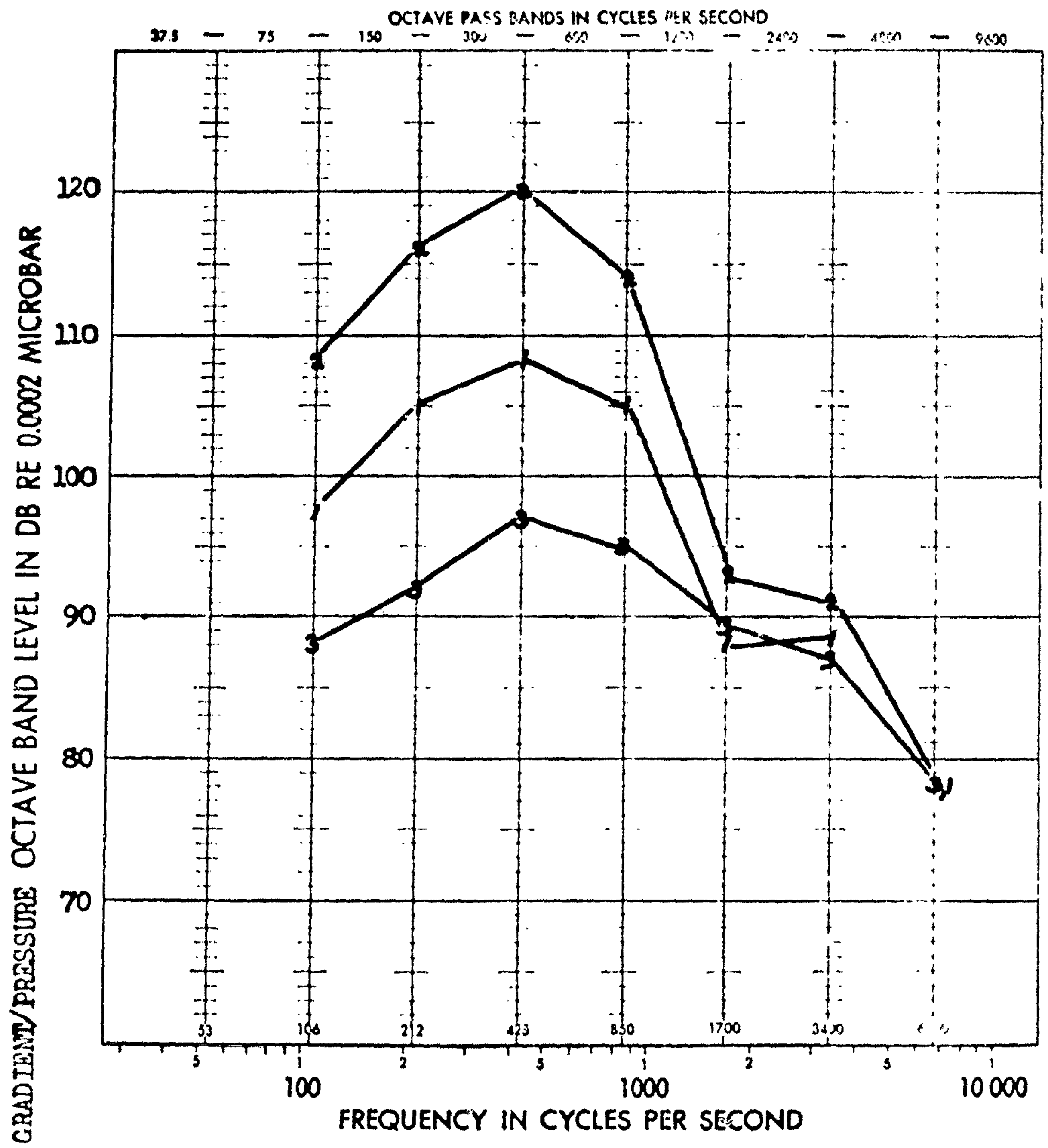
Curve 1: Noise cancellation in Fiberglas Noise Shield. Leak provided by 3" long 3/8" I.D. tube.

Cuuve 2: Noise cancellation in open diffuse noise field at dummy's lips.

Note: Western Electric 640AA probe microphones used. Measurements taken using dummy head; similar results occur on human subjects.

Figure A4-35

LONG TIME AVERAGE SPEECH SPECTRUM USING GRADIENT MICROPHONE IN WEAL FIBERGLAS NOISE SHIELD



Curve 1: Gradient microphone in noise shield.

Curve 2: Pressure Microphone in noise shield.

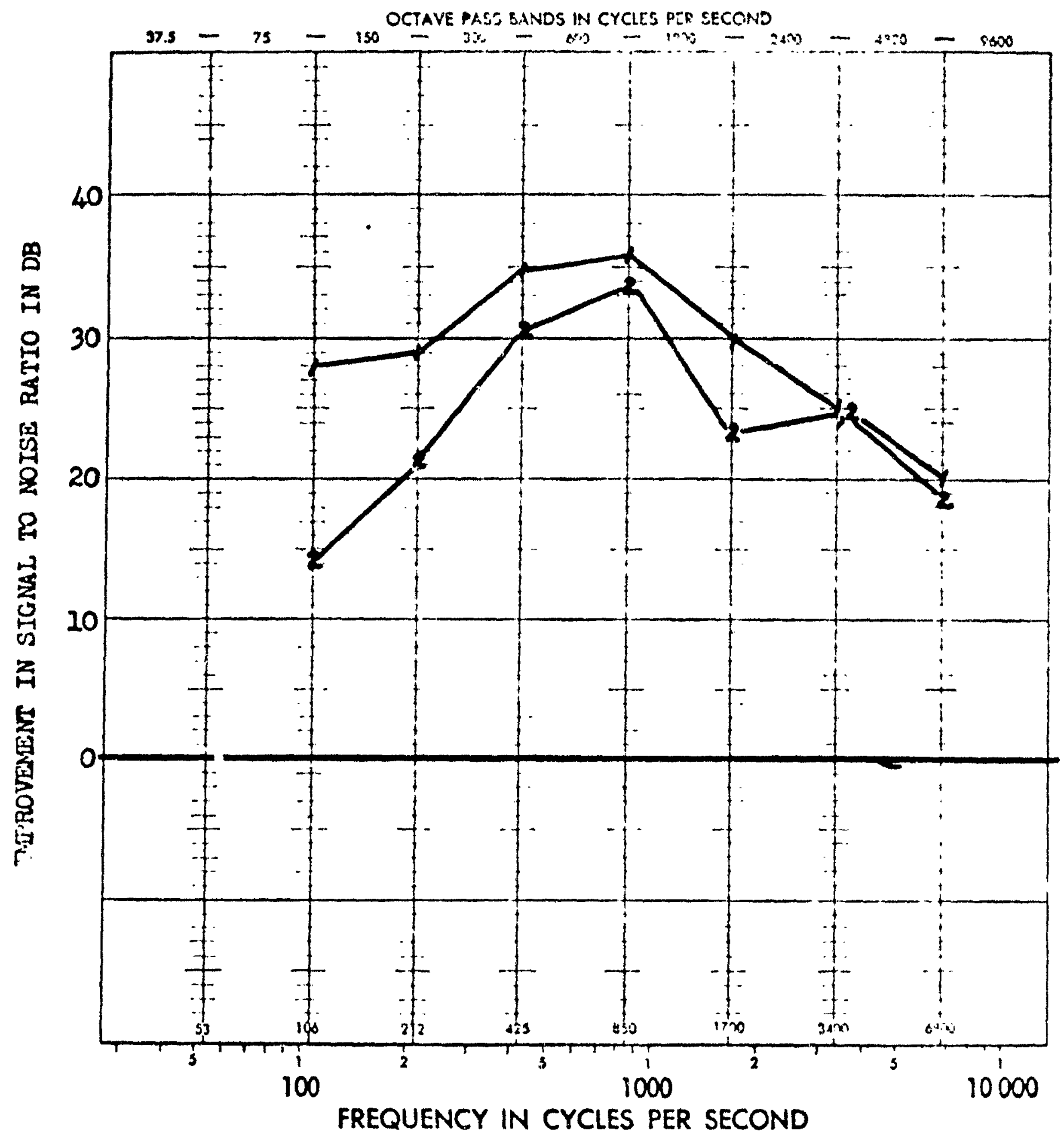
Curve 3: Pressure microphone in open at the lips.

Note: This is typical for all subjects.
Western Electric 640AA probe microphones used.

Same speaking effort for all results

Figure A4-36

IMPROVEMENT IN SIGNAL TO NOISE RATIO OF WEAL
GRADIENT MICROPHONE IN FIBERGLAS NOISE SHIEID
RELATIVE TO THE REFERENCE SYSTEM. SPEAKER: TW



Curve 1: WEAL 640AA probe tube gradient microphone in WEAL fiberglas noise shield.

Curve 2: WEAL 640AA probe tube pressure microphone in WEAL fiberglas noise shield.

Note: Leak provided by 3" long 3/8" I.D. tube.
Reference system is a Western Electric 640A" pressure probe microphone at the lips in the open.

Figure A4-37

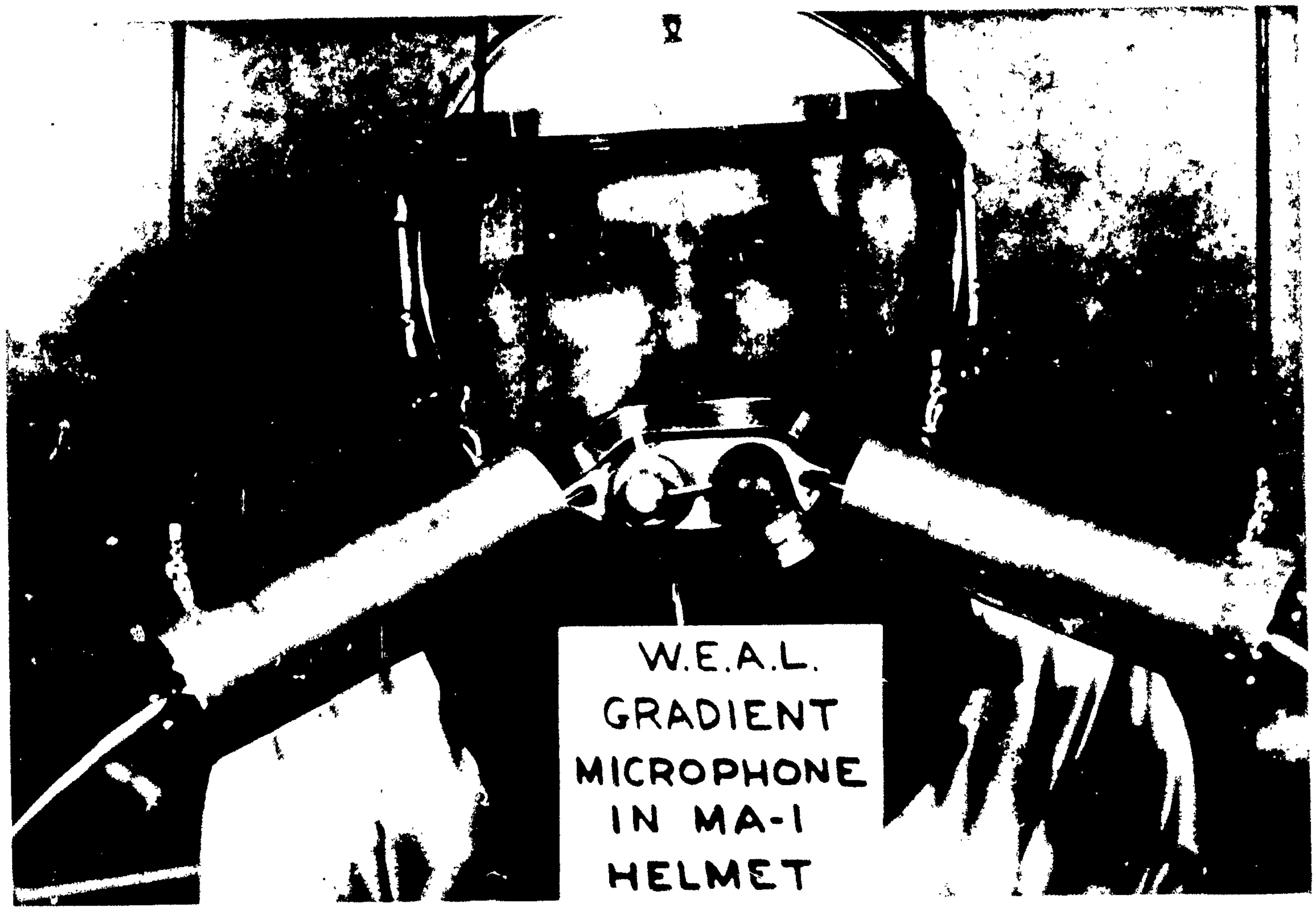
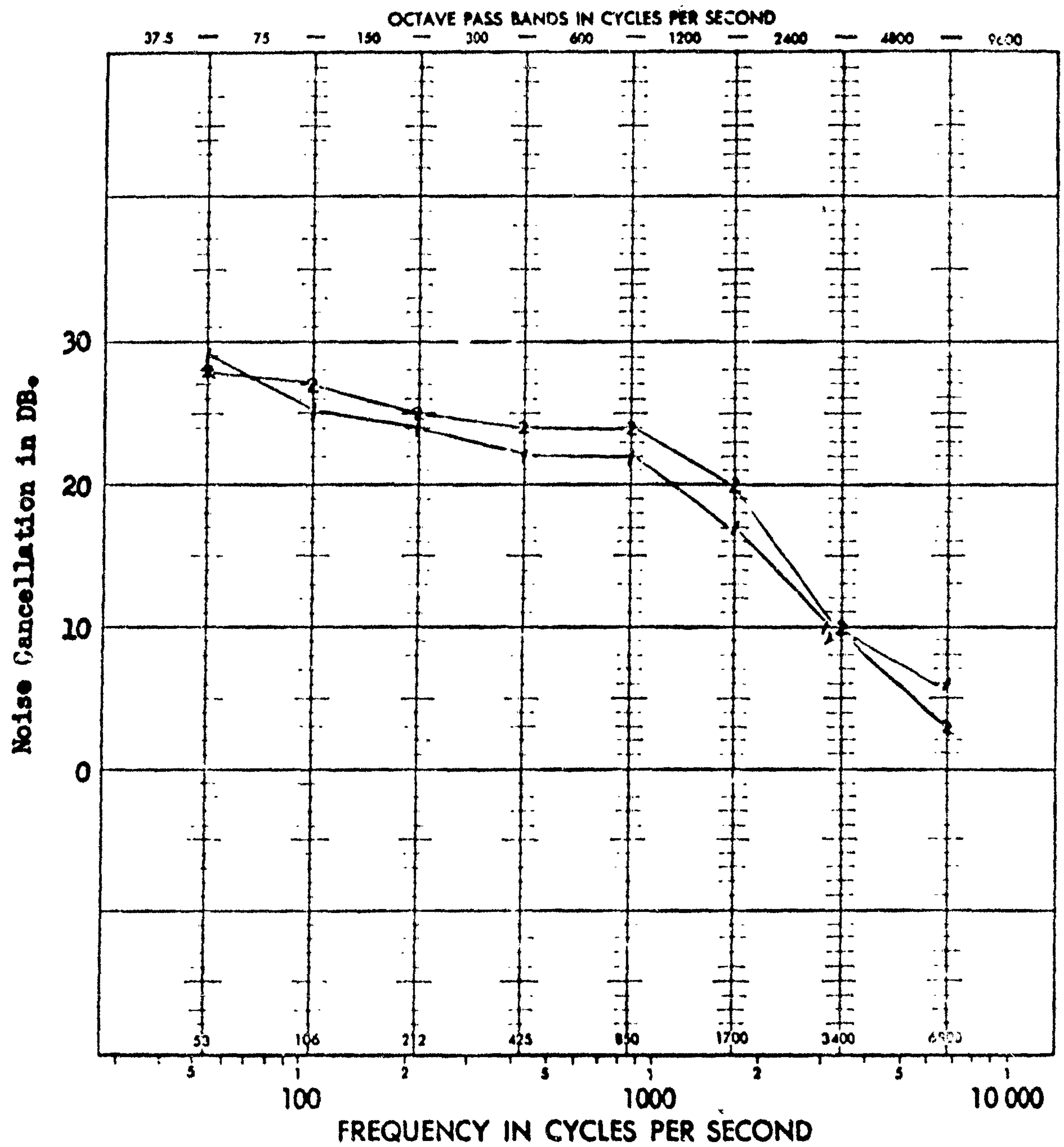


Figure A4 38

NOISE CANCELLATION OF GRADIENT MICROPHONE IN MA-1 HELMET ON DUMMY HEAD.

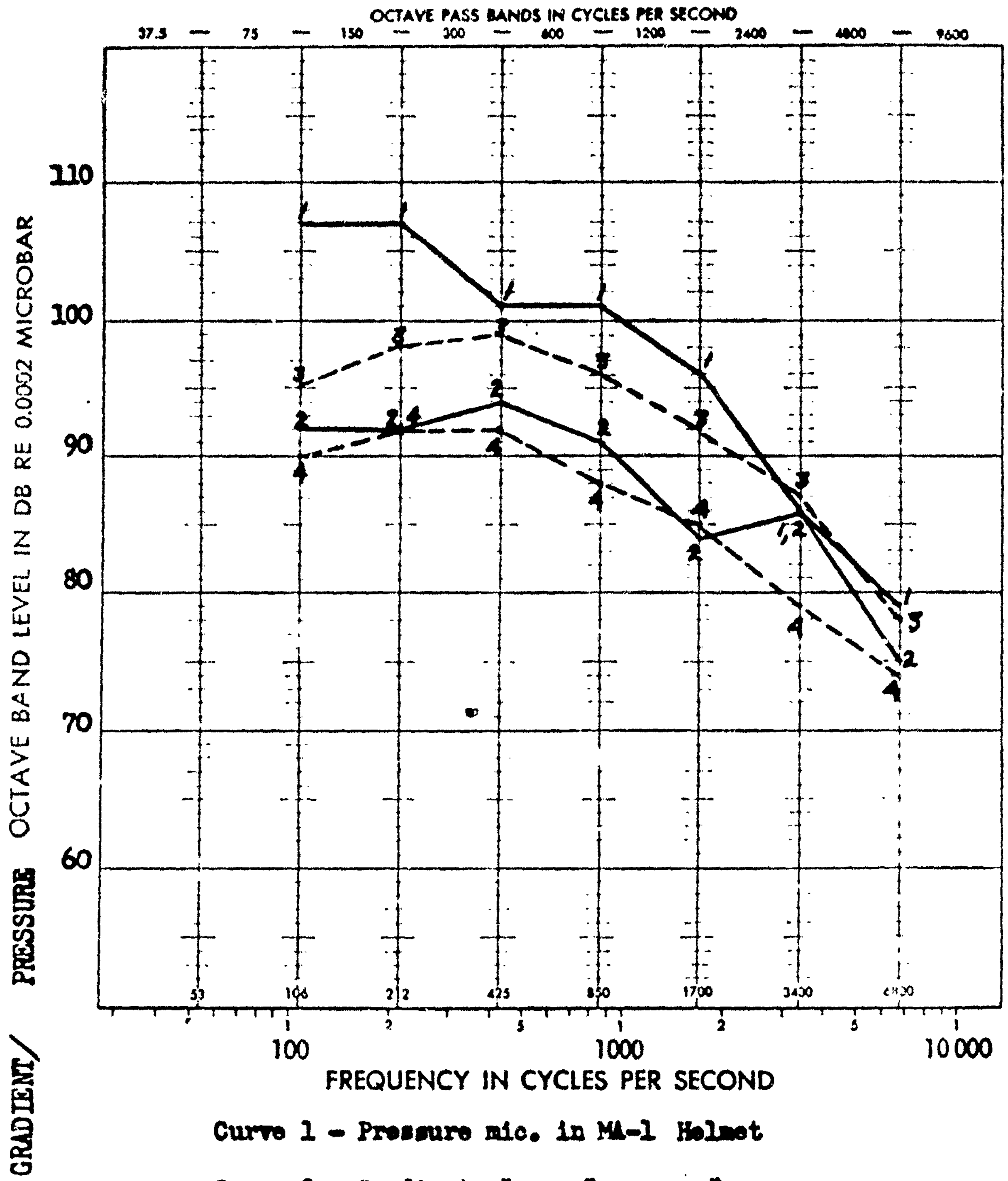


Curve 1: Noise cancellation in open diffuse noise field at dummy's lips.

Curve 2: Noise cancellation of gradient microphone in MA-1 helmet on dummy's head.

Figure A4-39

Long time average Speech Spectra in open and in MA-1 Helmet. Subject TW.



Curve 1 - Pressure mic. in MA-1 Helmet

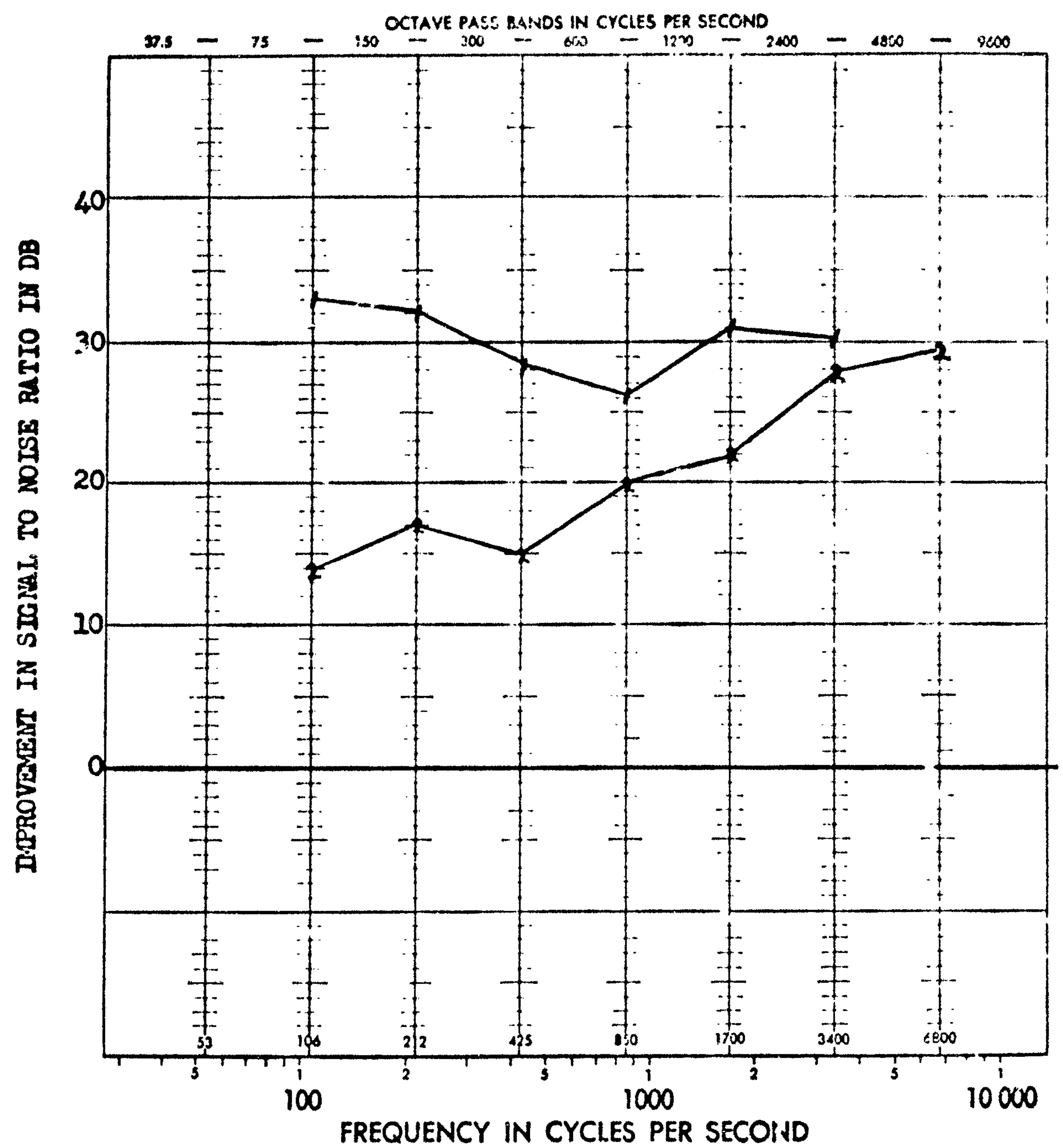
Curve 2 - Gradient " " "

Curve 3 - Pressure " open

Curve 4 - Gradient " open

Figure A4-40

IMPROVEMENT IN SIGNAL TO NOISE RATIO OF WEAL GRADIENT MICROPHONE IN MA-1 HELMET RELATIVE TO THE REFERENCE SYSTEM. SPEAKER: TW.

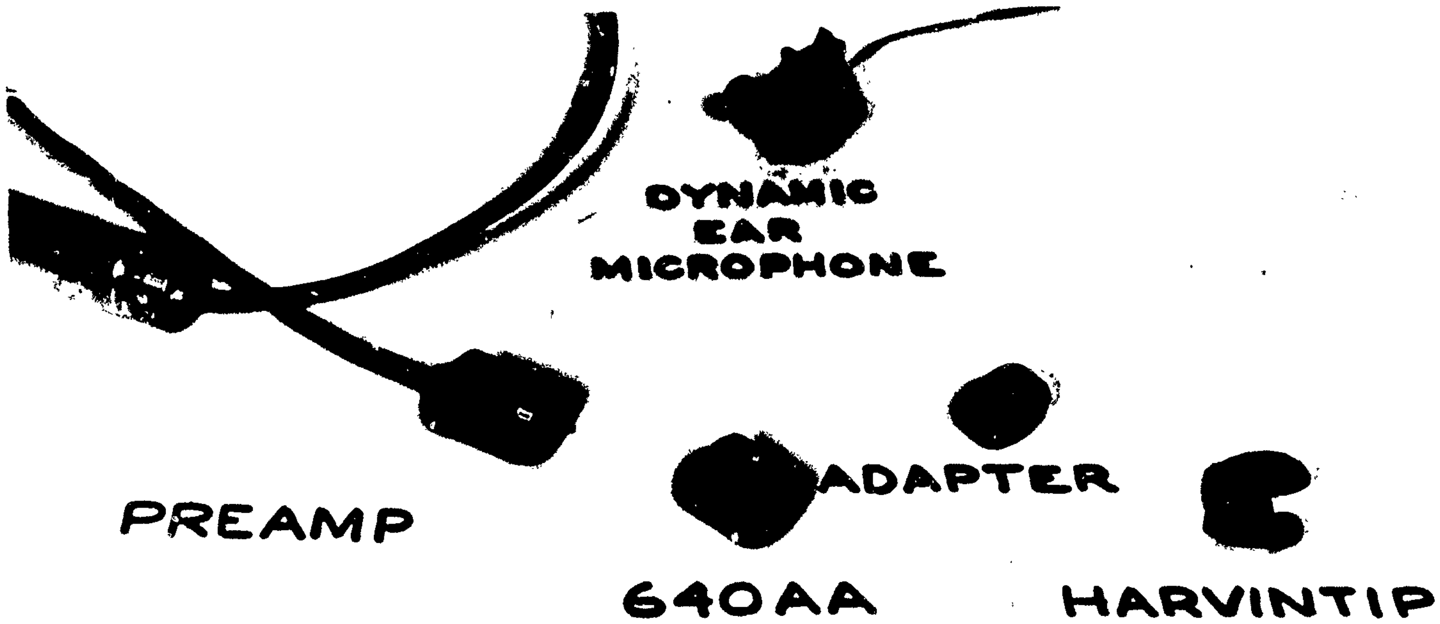


Curve 1: WEAL probe tube gradient microphone in MA-1 Helmet.

Curve 2: WEAL probe tube pressure microphone in MA-1 Helmet.

Note: Reference system is a Western Electric 640AA pressure probe microphone at the lips in the open.

Figure A4-41



EAR
MICROPHONE
USING 640AA

RELATIVE CALIBRATION OF 640 AA EAR MICROPHONE

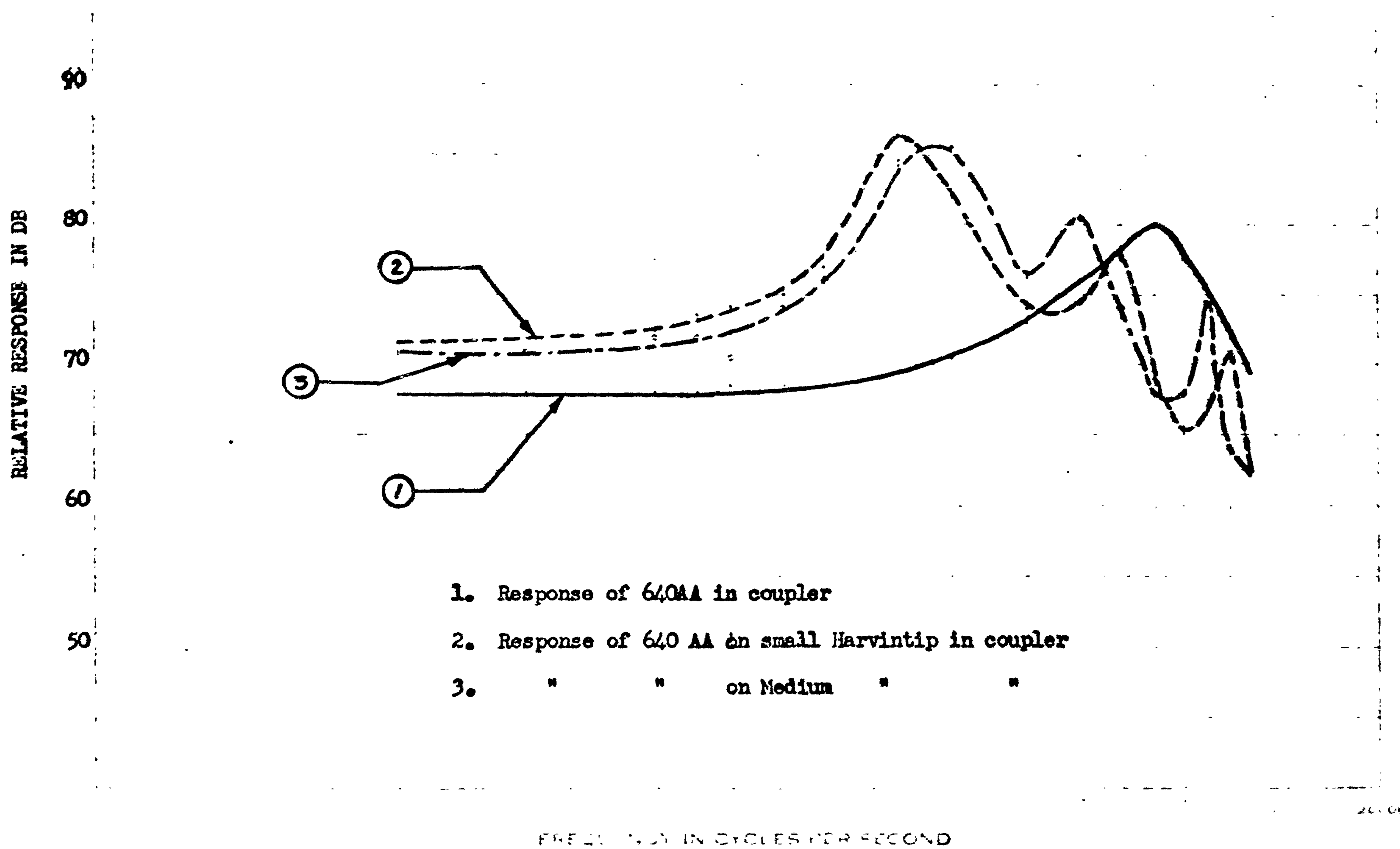
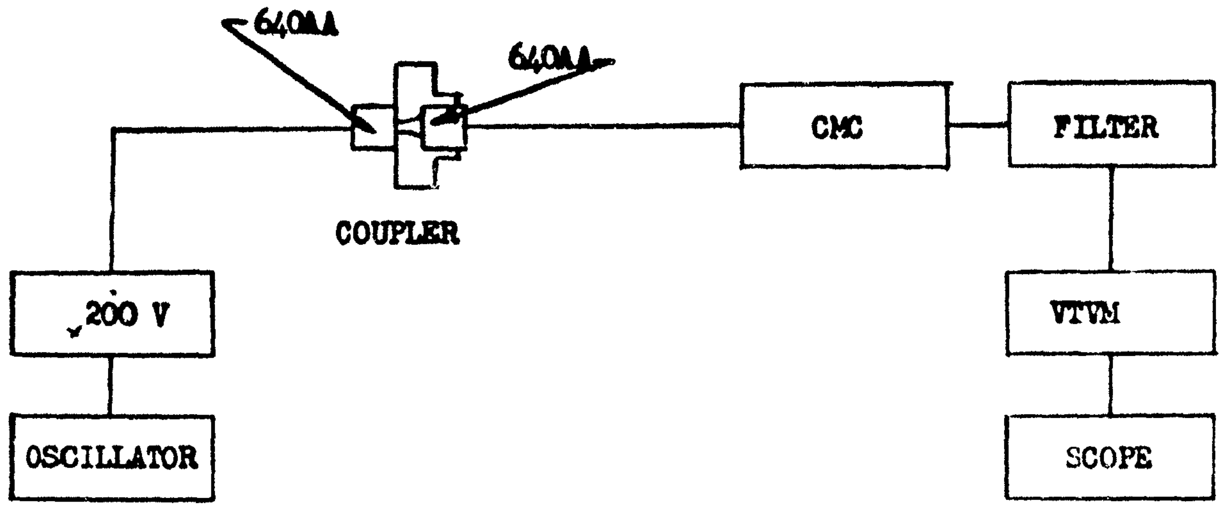
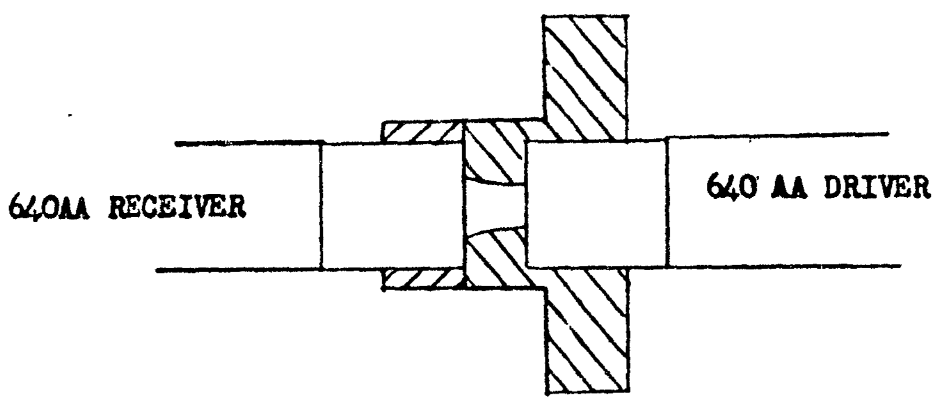


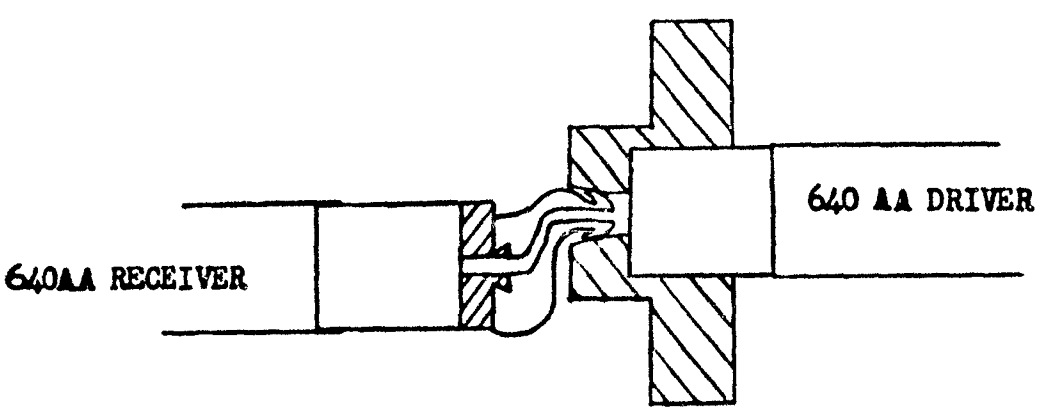
Figure A4-43

APPARATUS USED FOR CALIBRATING EAR MICROPHONES



MEASURING RESPONSE OF 640 AA IN COUPLER



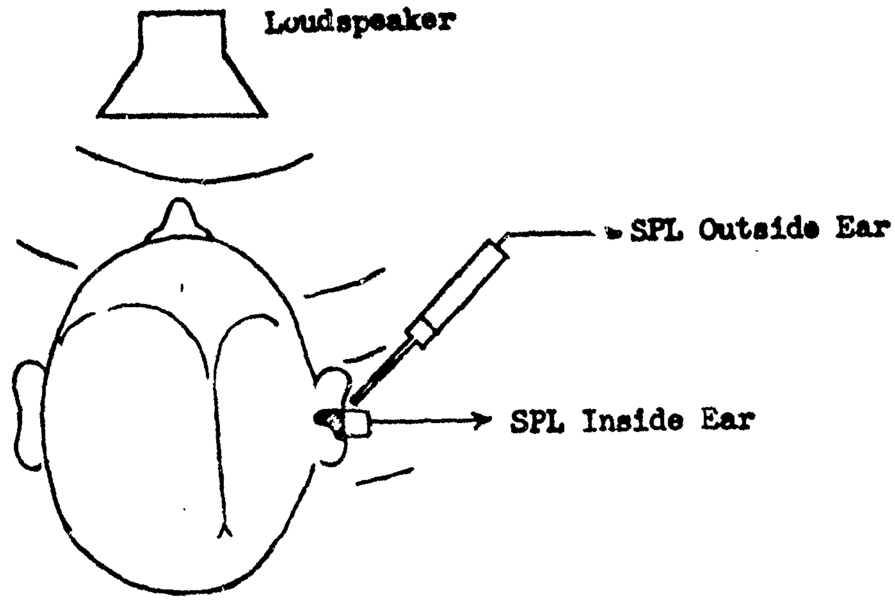


MEASURING RESPONSE OF HARVINTIP
WITH 640 in COUPLER

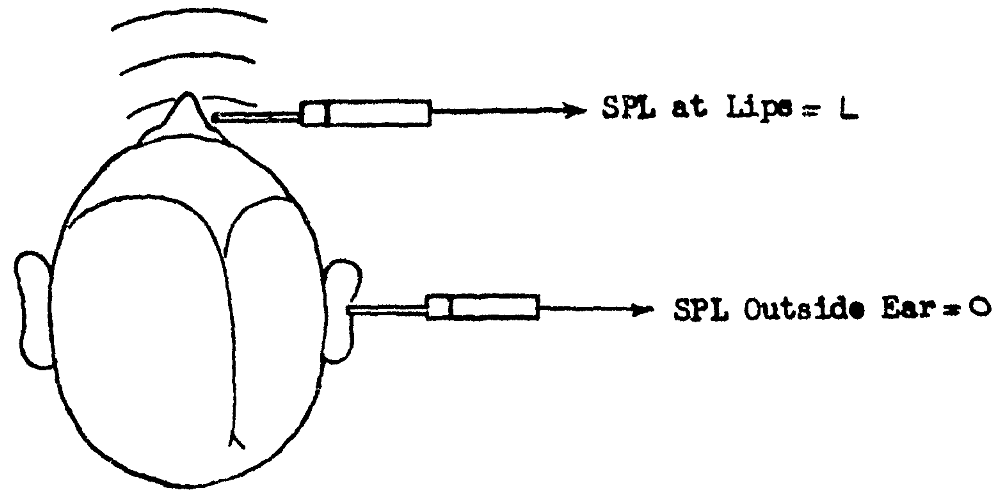
Figure A4-44

I. Measurement of Attenuation of Ear Pickup.

Attenuation in DB
SPL outside ear - SPL inside ear



II. Measurement of Speech Level Outside Ear



III. Measurement of Speech Level Inside Ear

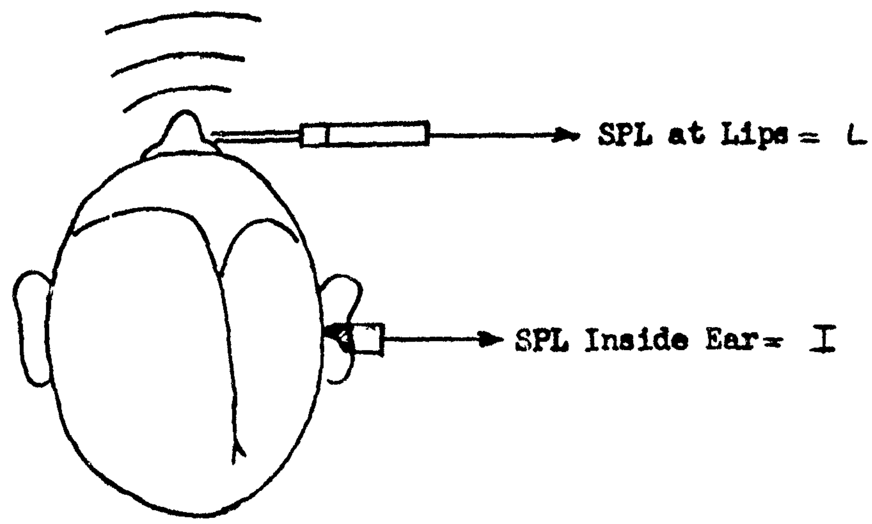
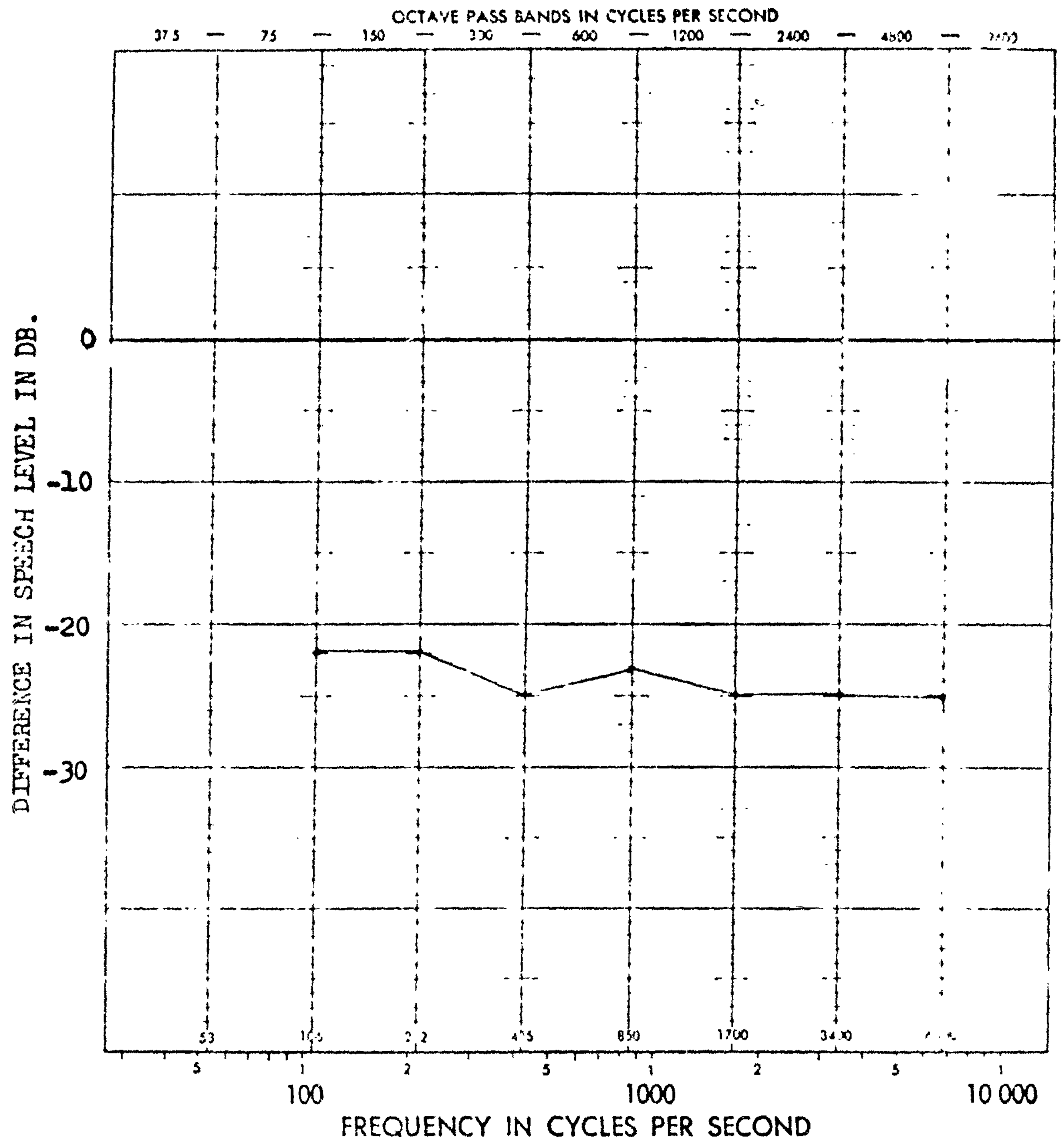


Figure A4-46

REDUCTION IN SPEECH LEVEL FROM THE LIPS TO JUST OUTSIDE THE EAR.



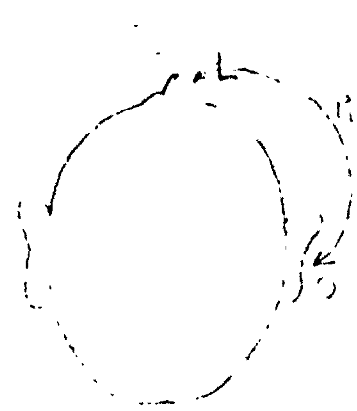
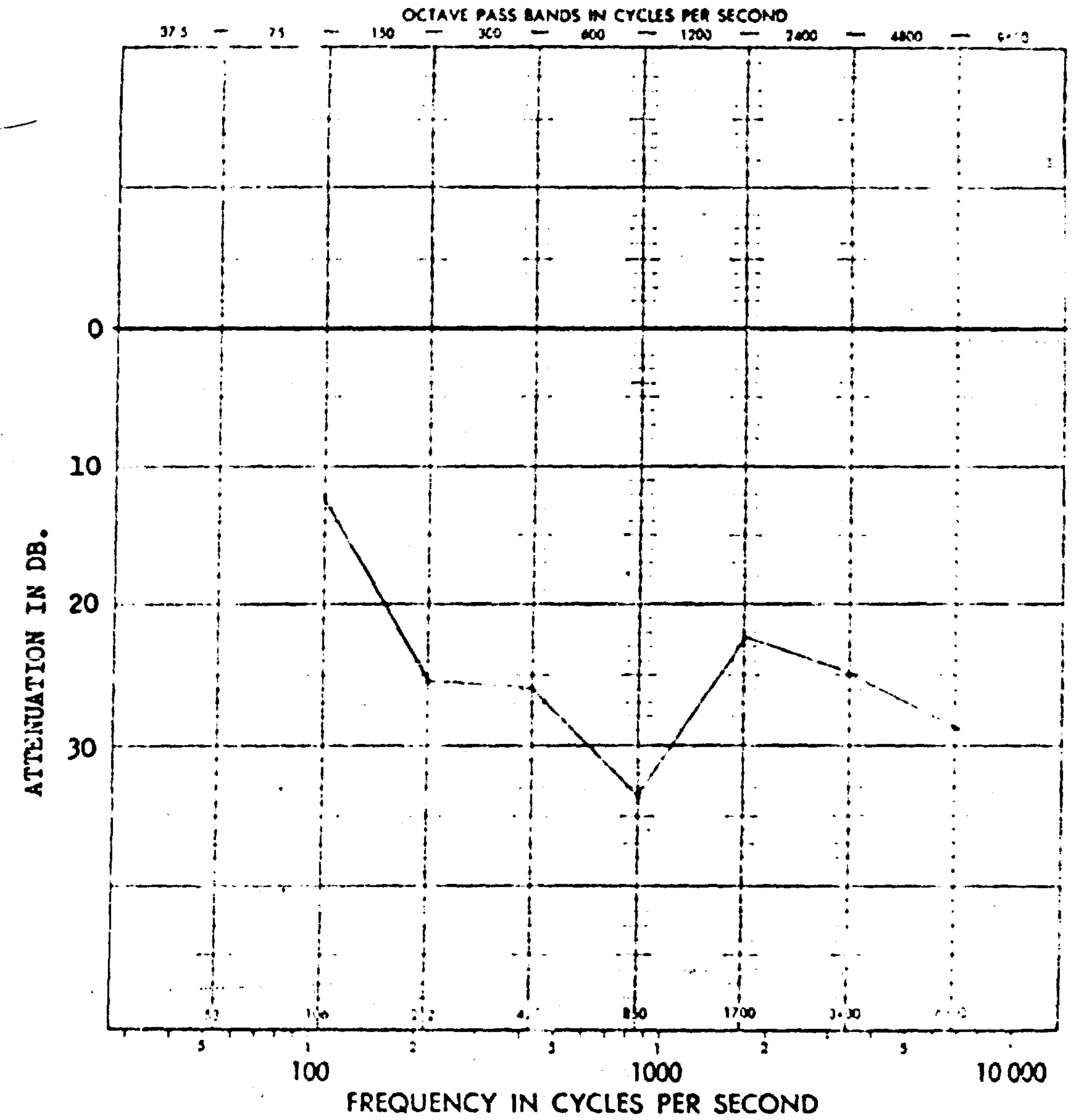


Figure A4-47

ATTENUATION OF HARVINTIP + 640AA ASSEMBLY TO AIRBORNE SOUND.



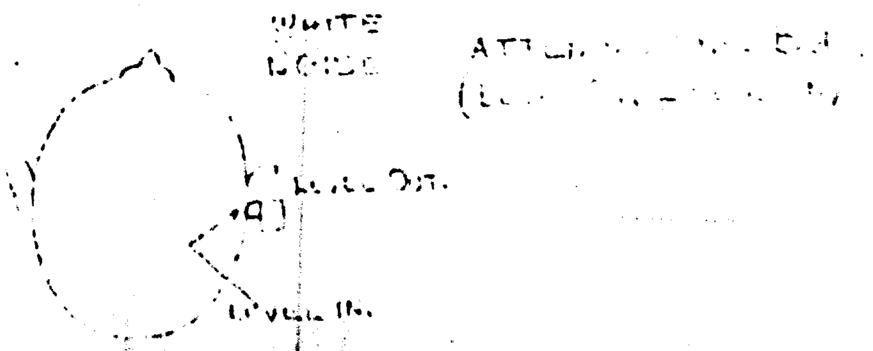
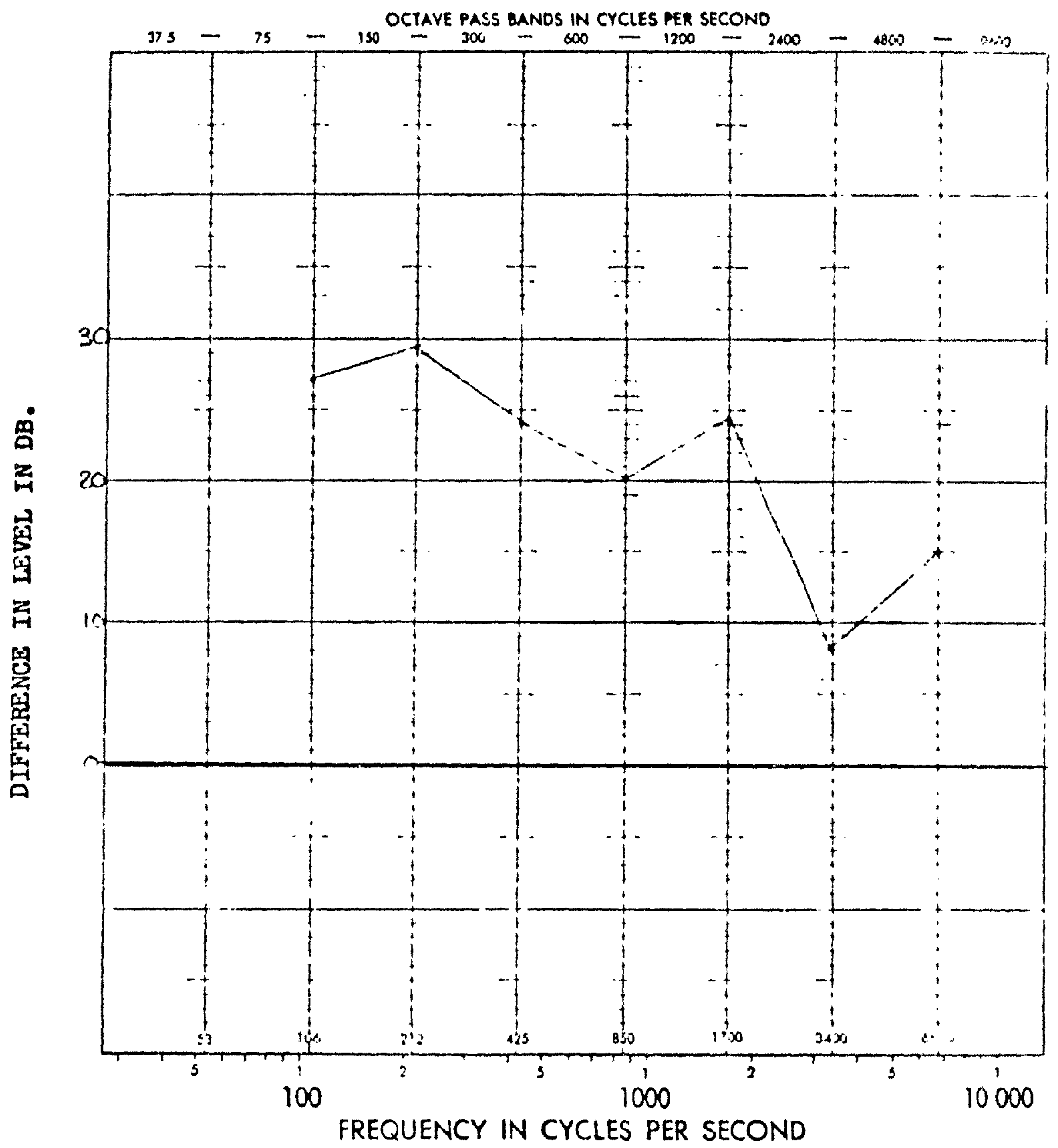


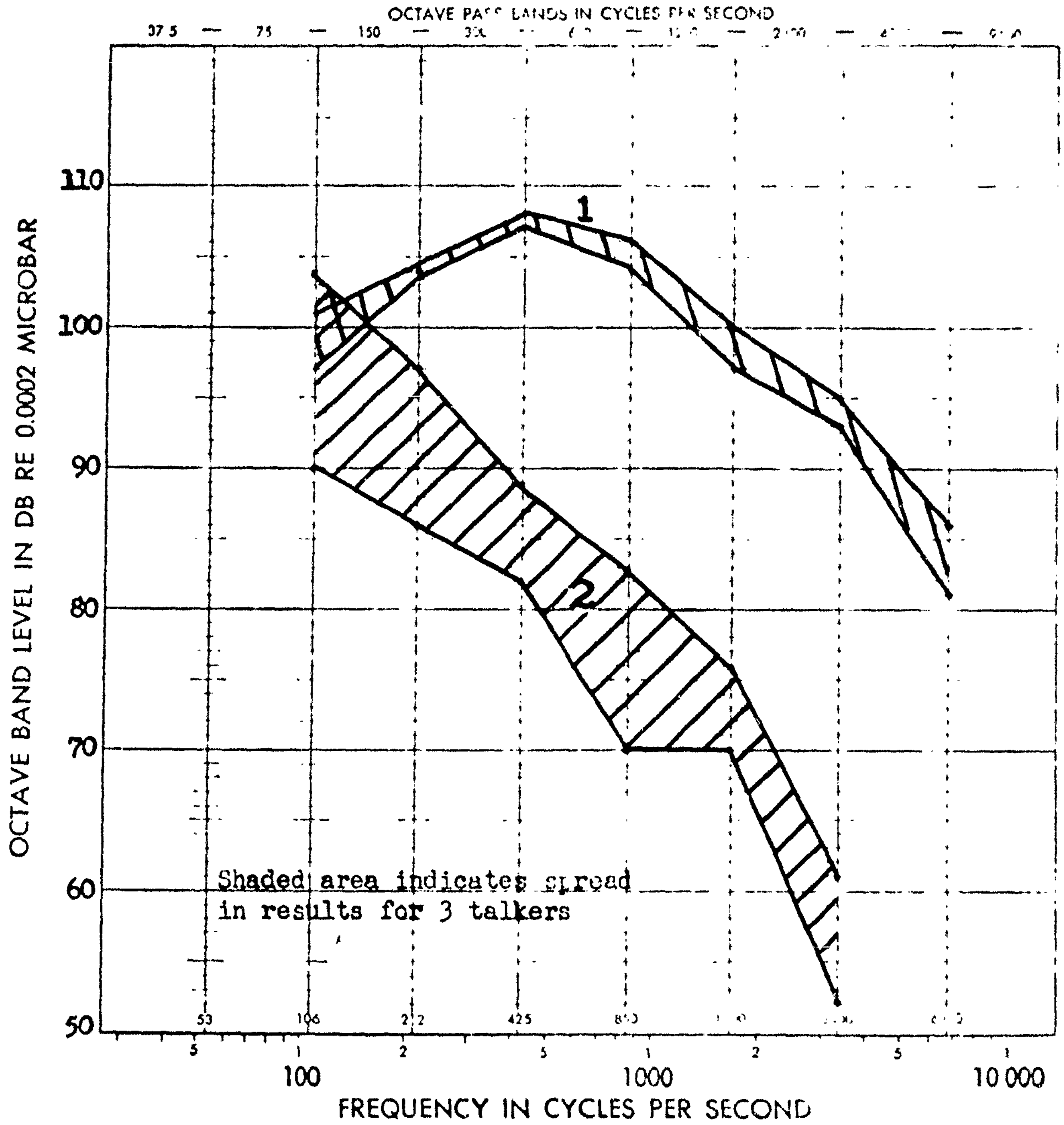
Figure A4-48



Difference in level between Bone Conducted + Airborne Speech, and Airborne speech only. Subject T.W with WE 640 AA in Harvintip.

Figure A4-49

LONG TIME AVERAGE SPEECH SPECTRUM IN THE EAR AND AT THE LIPS.



Curve 1: LTA speech spectrum as measured by Reference System (W.E. 640AA pressure probe microphone at lips). All spectra are 110 DB over-all SPL.

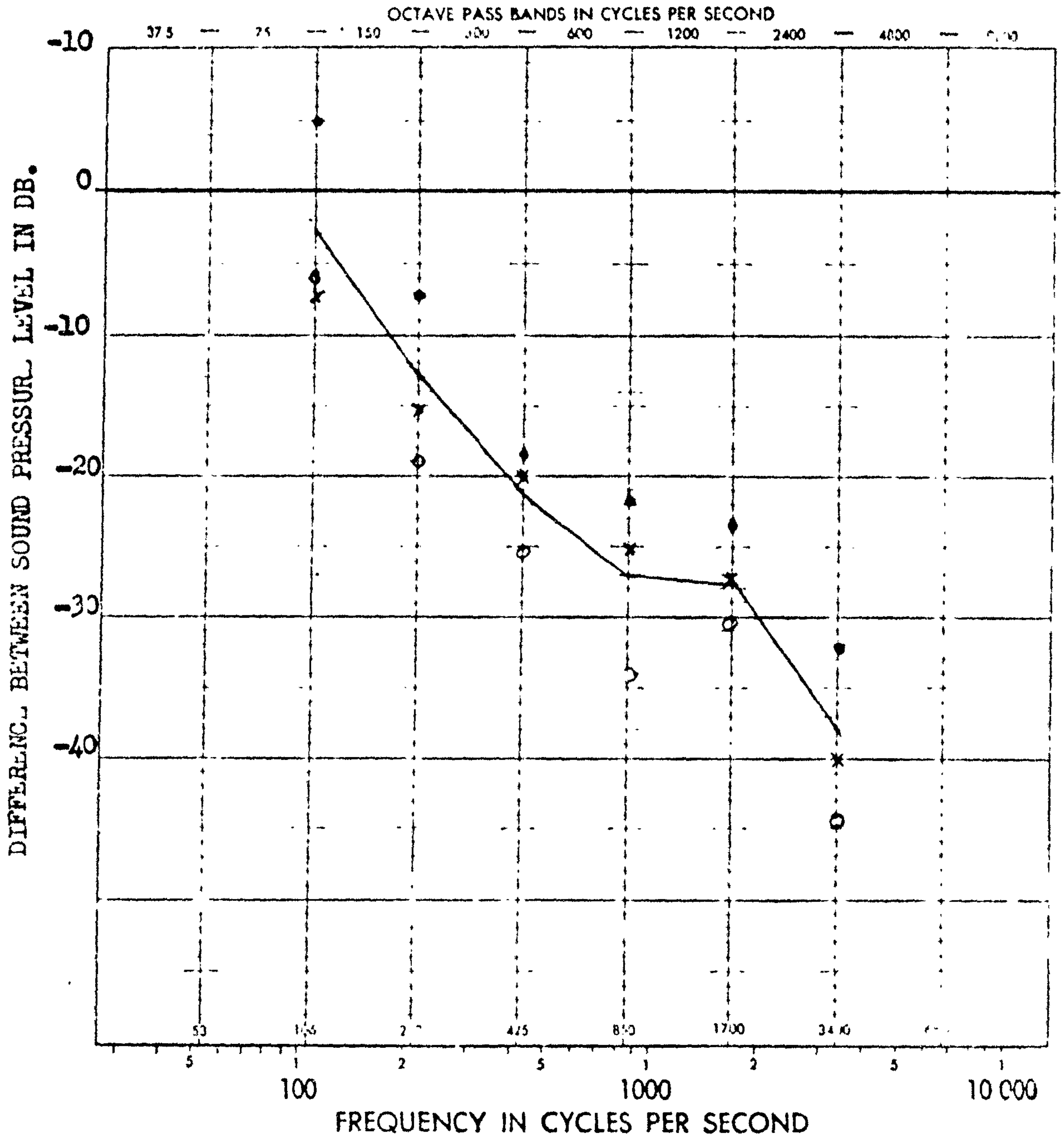
Curve 2: LTA as measured by W.E. 640AA in Harvintip insert device closely coupled to ear canal.

Note: Same speaking level for both conditions.

Subjects: WO, JPC, TW

Figure A4-50

DIFFERENCE BETWEEN SOUND PRESSURE LEVEL IN EAR AND AT THE LIPS FOR SAME SPEAKING LEVEL.



The curve represents average of data on 3 subjects.

Subjects: ● WO

x JPC

O TW

Figure A4-51

COMPLIMENTARY EAR TRANSDUCER RESPONSE
REQUIRED TO MATCH THE SPEECH SPECTRUM
AT THE LIPS.

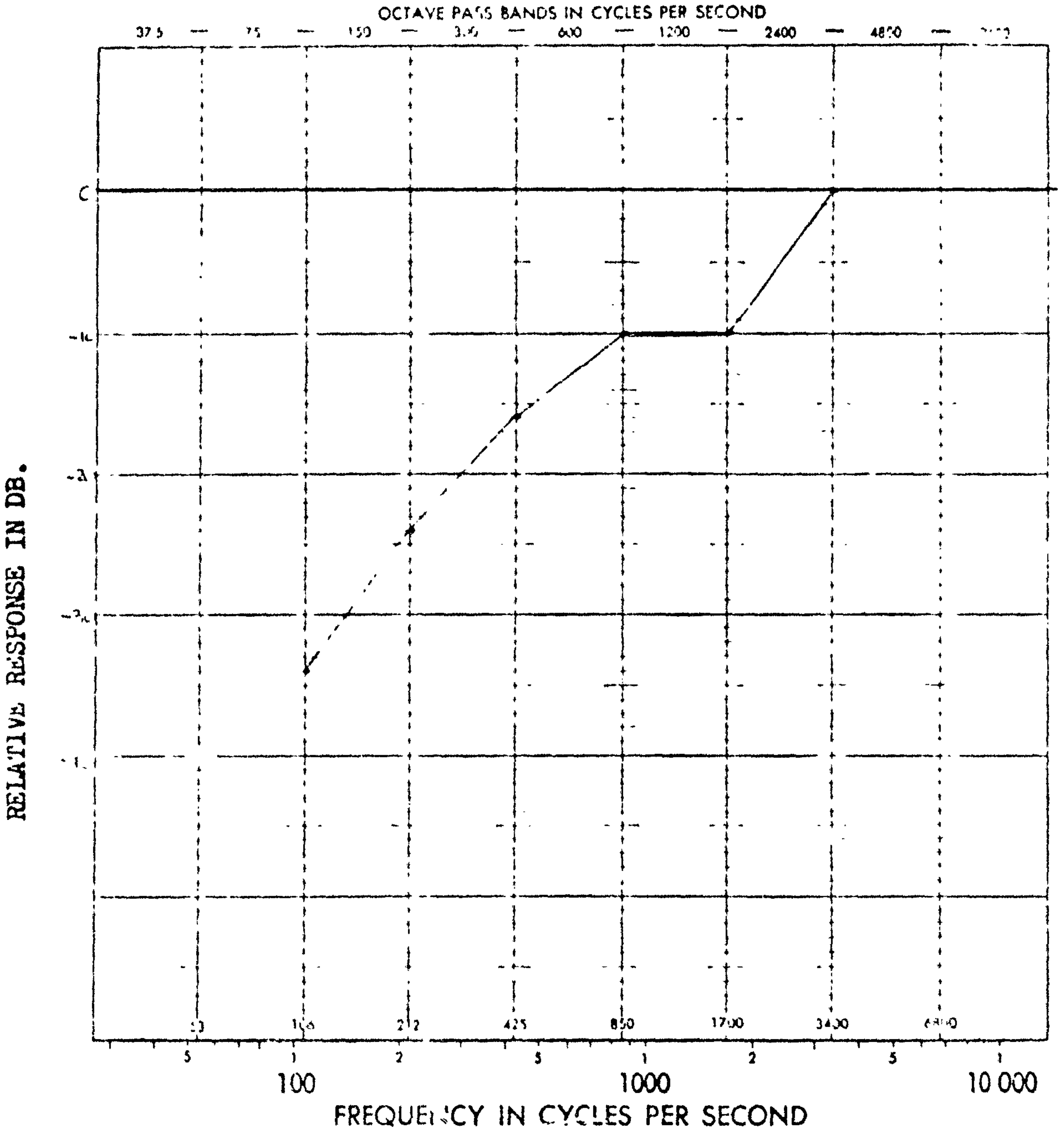
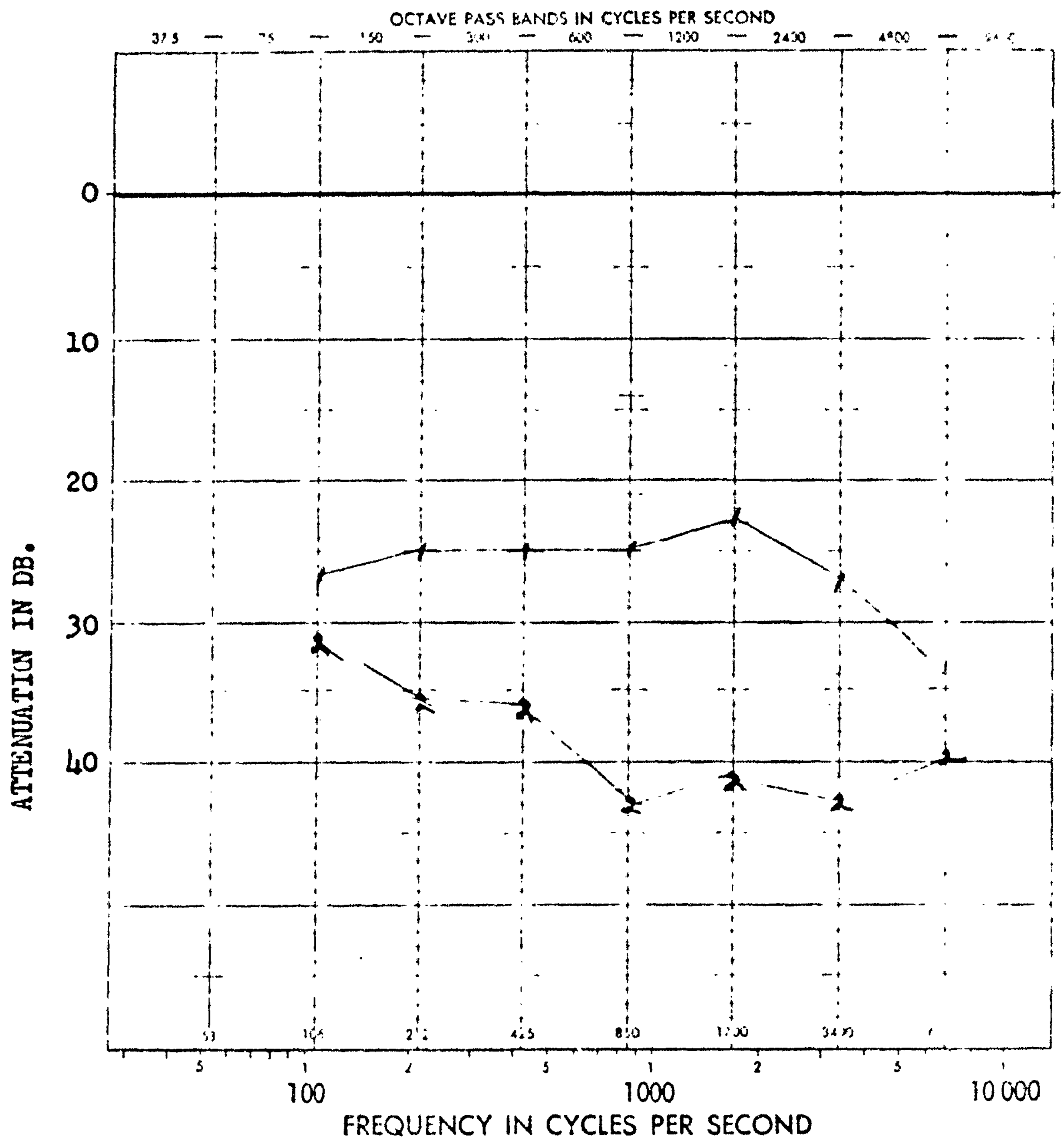


Figure A4-52

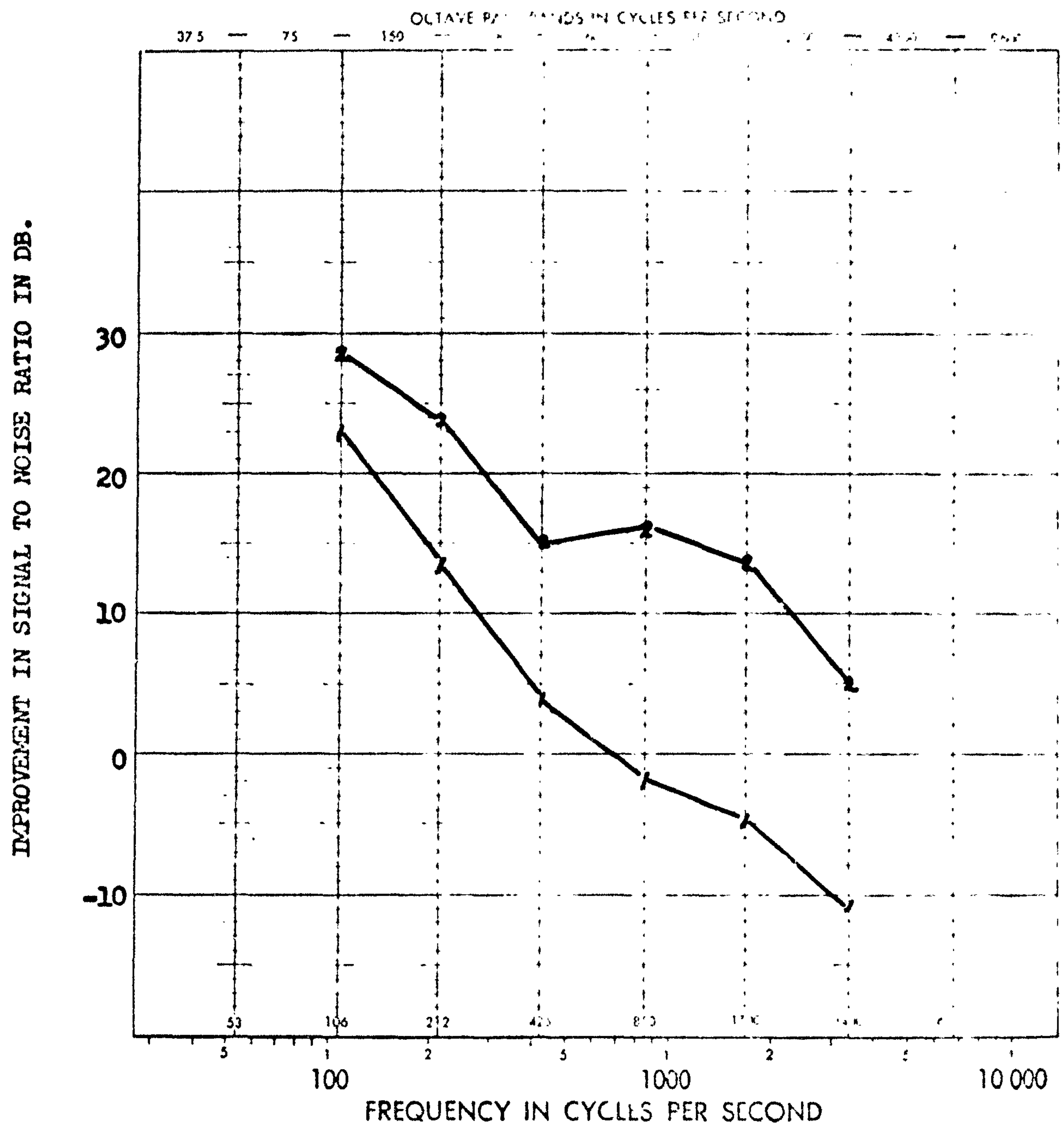
ATTENUATION OF NOISE EXCLUDING DEVICES WHICH COULD BE USED WITH AN EAR PICKUP.



Curve 1: Noise attenuation provided by Harvintip.
Curve 2: Noise attenuation provided by earplug + muff.
Curve 3: Noise attenuation provided by earplug + helmet (data not available at this time)

Figure A4-53

THE IMPROVEMENT IN SIGNAL TO NOISE RATIO OVER AN OPEN PRESSURE MICROPHONE AT THE LIPS USING A 640AA EAR MICROPHONE WITH SEVERAL NOISE EXCLUDING DEVICES



curve 1: Ear microphone with Harvintip only

curve 2: Ear microphone with Harvintip + earmuff (estimated)

Figure A4-54


DYNAMIC
EAR
MICROPHONE

RELATIVE CALIBRATION OF DYNAMIC EAR MICROPHONE

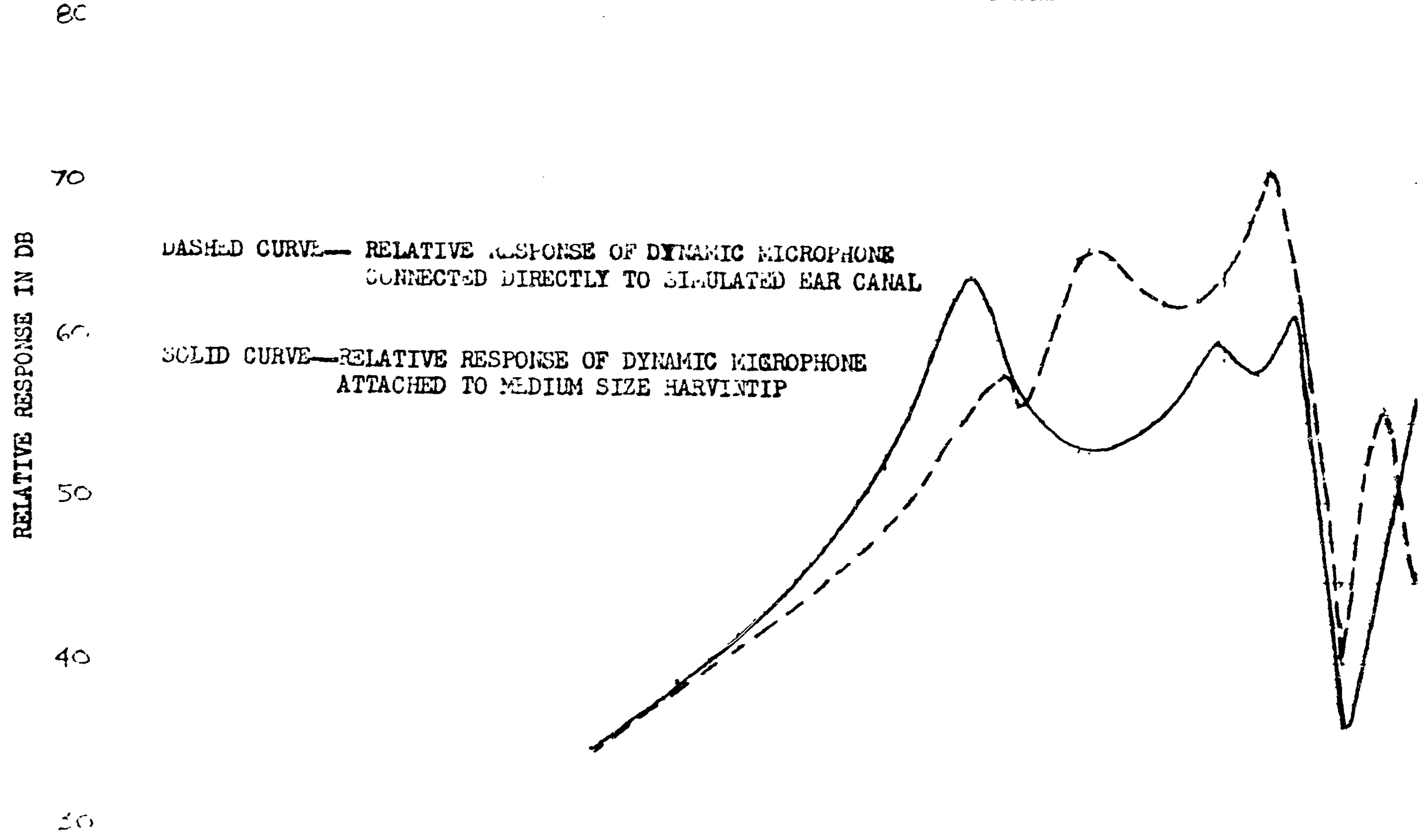
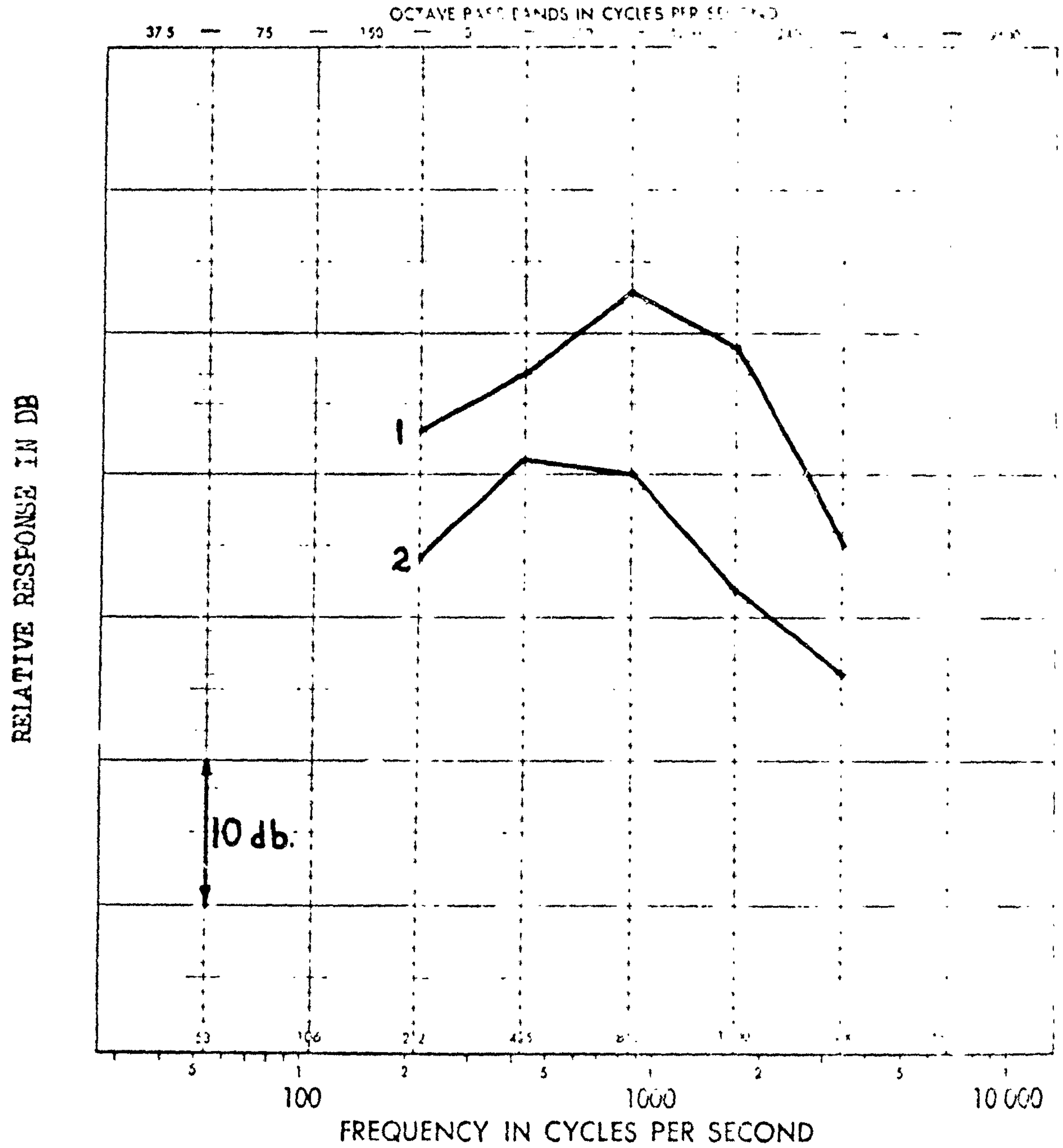


Figure A4-56

LONG TIME AVERAGE SPEECH SPECTRA USING DYNAMIC EAR MICROPHONE.

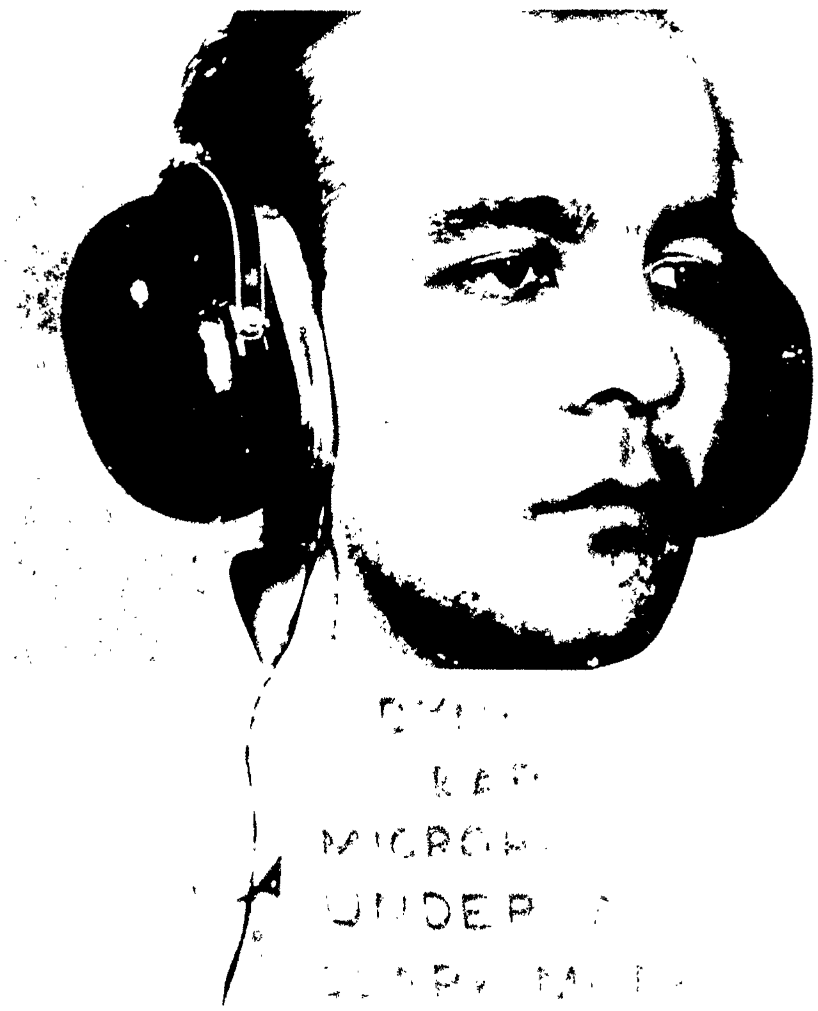


Curve 1: Dynamic ear microphone.

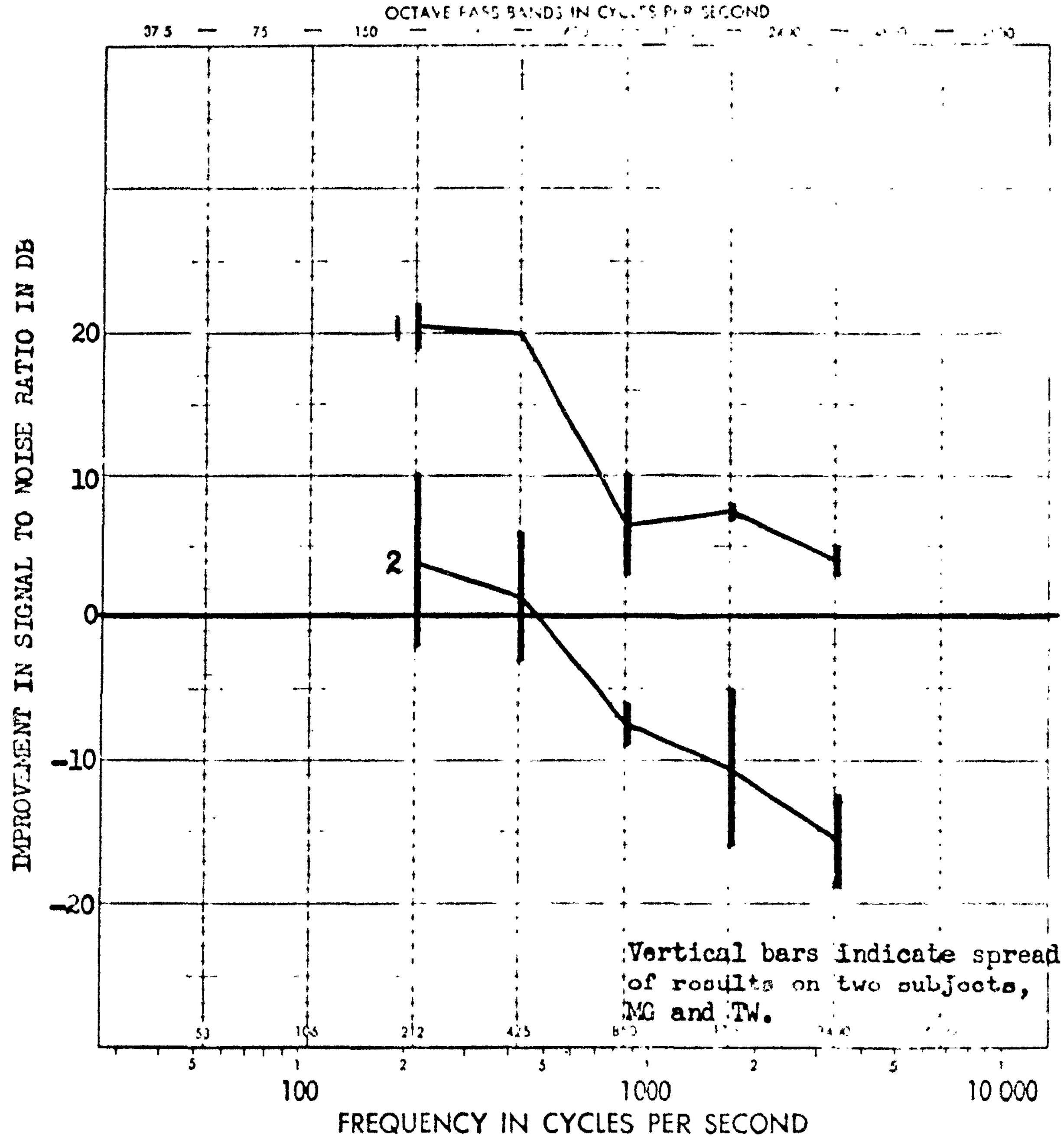
Curve 2: Reference System (W.E. 640AA pressure probe microphone at the lips in the open).

Note: Approximately "normal conversational level" for all spectra.

Figure A4-57



IMPROVEMENT IN SIGNAL TO NOISE RATIO OF DYNAMIC EAR MICROPHONE RELATIVE TO THE REFERENCE SYSTEM.



Curve 1: Dynamic ear microphone coupled to Harvintip under D. Clark Muff type.

Curve 2: Dynamic ear microphone coupled to Harvintip. No additional noise shielding.

Note: Reference System is W.E. 640AA pressure probe microphone at the lips in the open.

Figure A4-59

SIMPLE FOREHEAD PICKUP

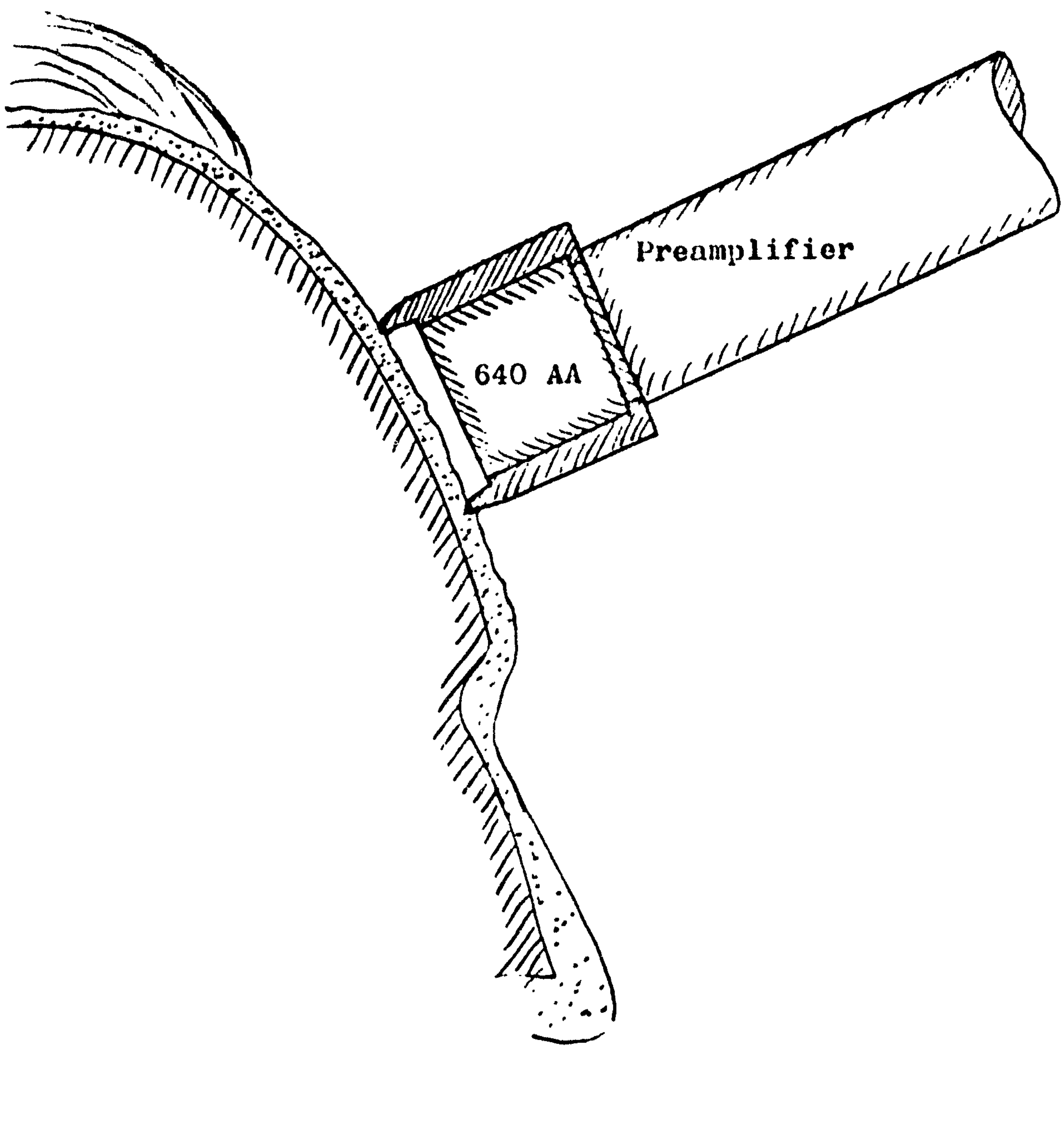
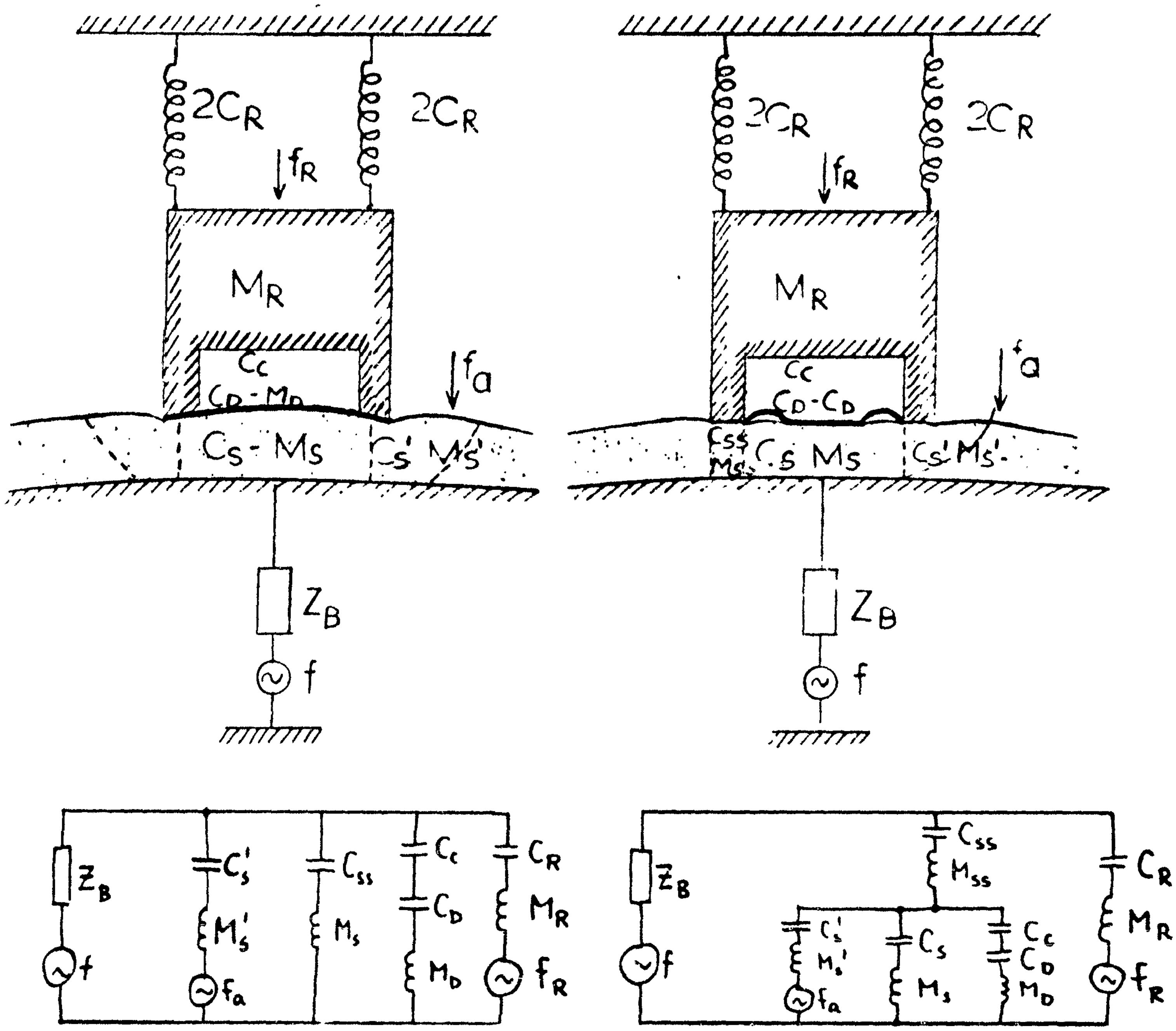
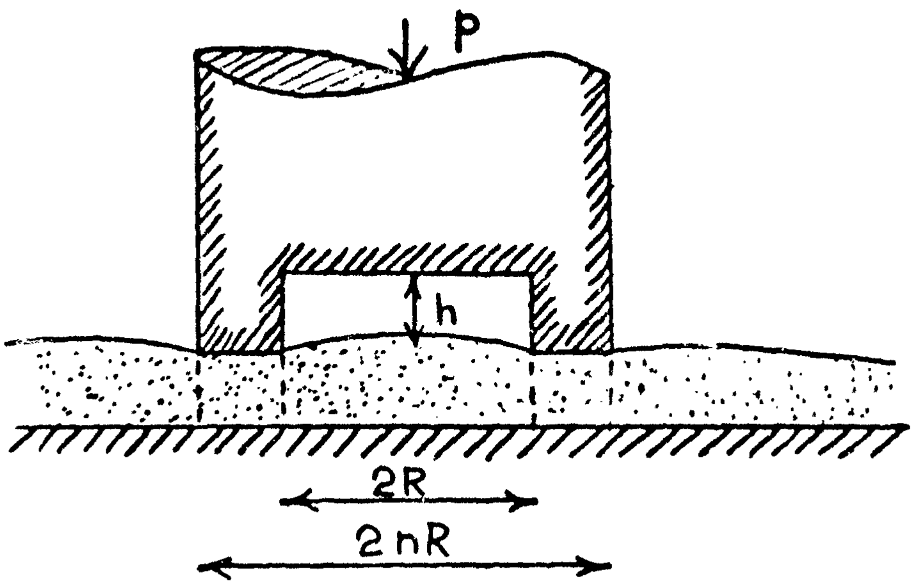


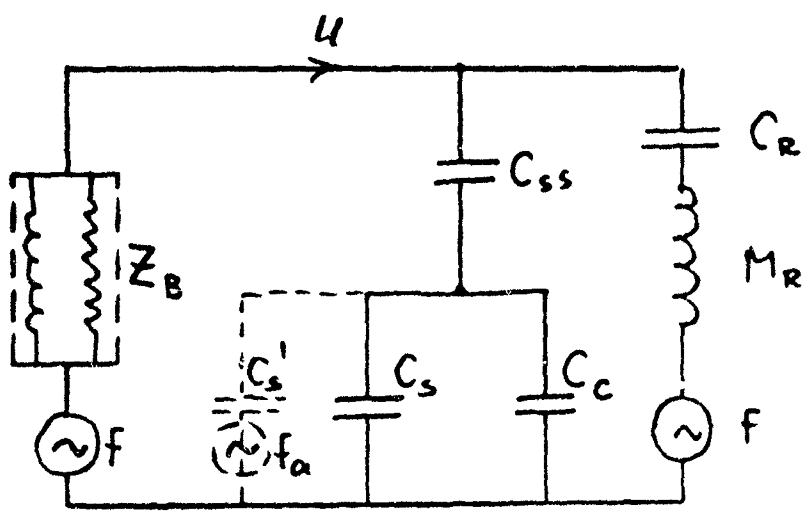
Figure A4-60



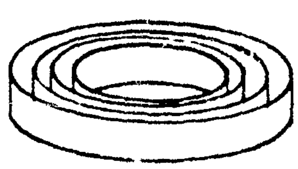
M_R = mass of couplerunit
C_R = compliance of coupler suspension
C_C = " " cavity
C_D, M_D = compliance and mass of diaphragm
C_s, M_s = " " of skin under diaphragm
C_{ss}, M_{ss} = " " " " " coupler edge
C_s', M_s' = " " " " for force f_a
Z_B = bone impedance
f = force due to internal vibrations (in throat and mouth cavity)
f_R = force on the coupler due to an external soundfield
f_a = force through the skin due to soundvibrations conducted.

Figure A4-61



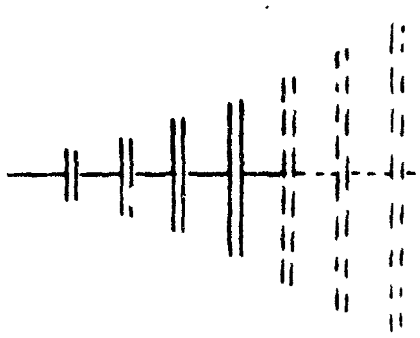


b



c

Mechanical representation of the transmission path under the coupler edge.

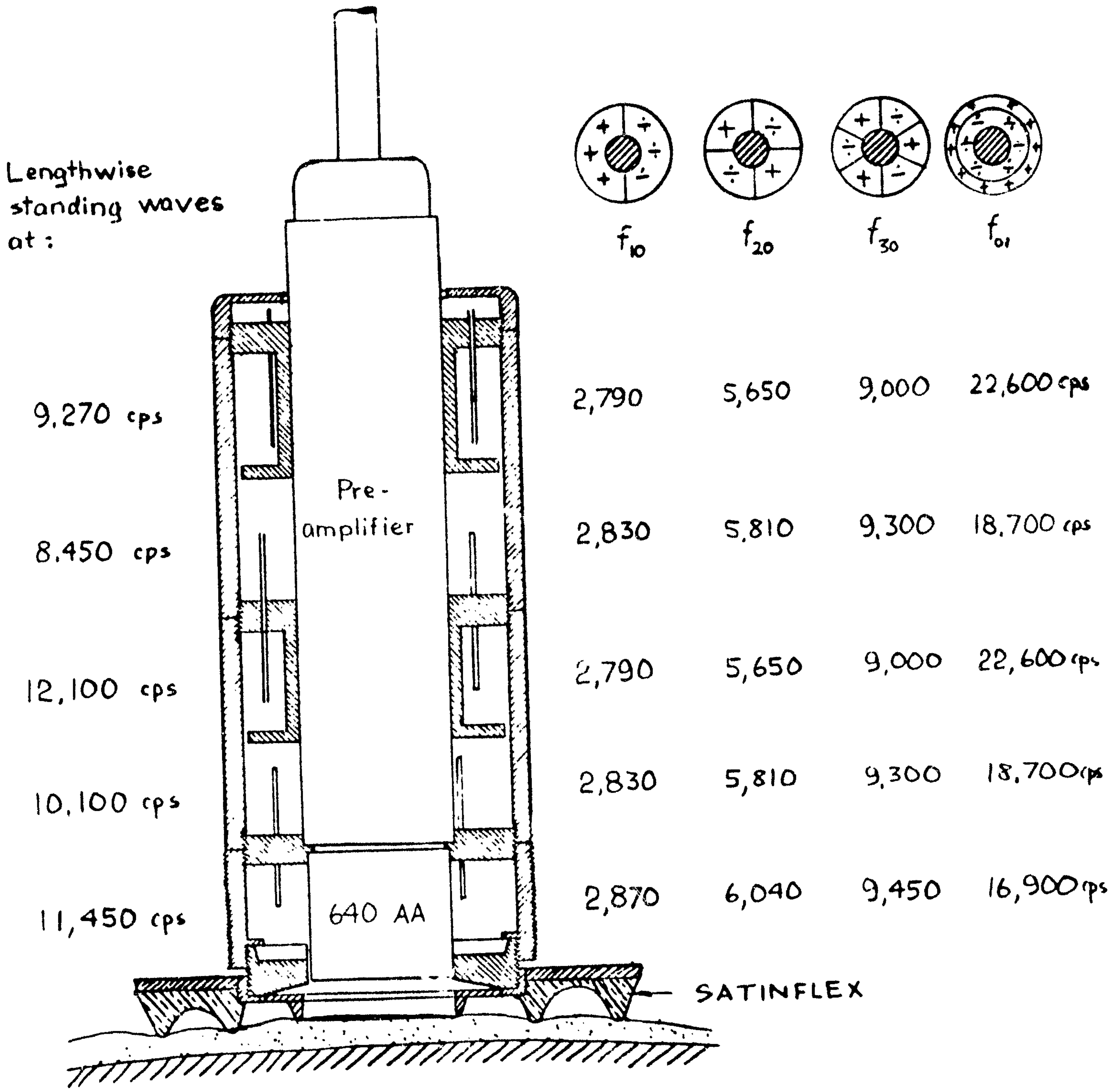


d

Electrical equivalent for c above.

Figure A4-62

FOREHEAD PICKUP
WITH FILTER

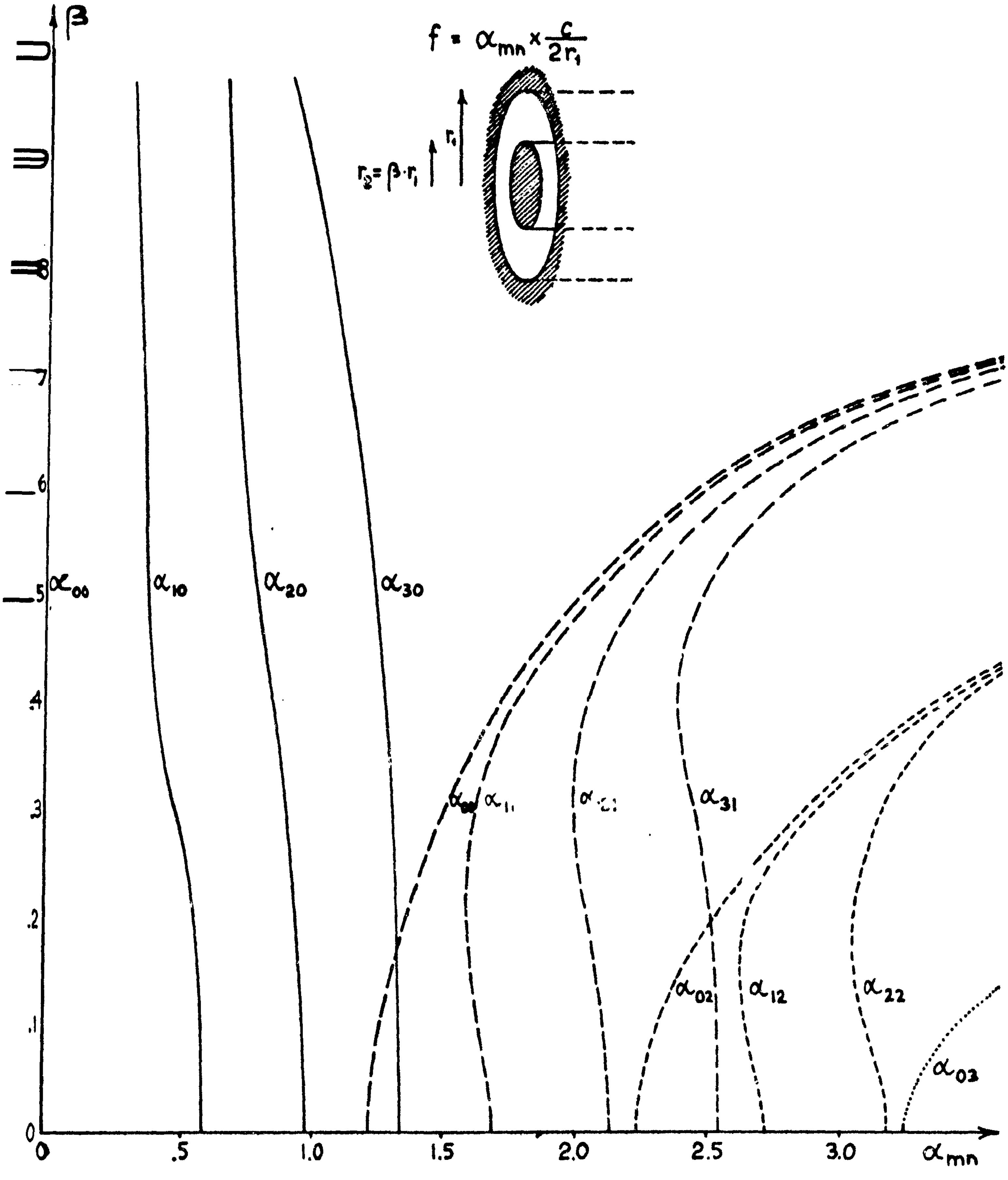


An acoustic filter for use with Western Electric Condenser Microphone 640 AA and Western Electro-Acoustic Laboratory preamplifier. Cavity resonances may be expected at the indicated frequencies.

Figure A4-63

WAVE TRANSMISSION INSIDE COAXIAL CYLINDERS
WITH RIGID WALLS

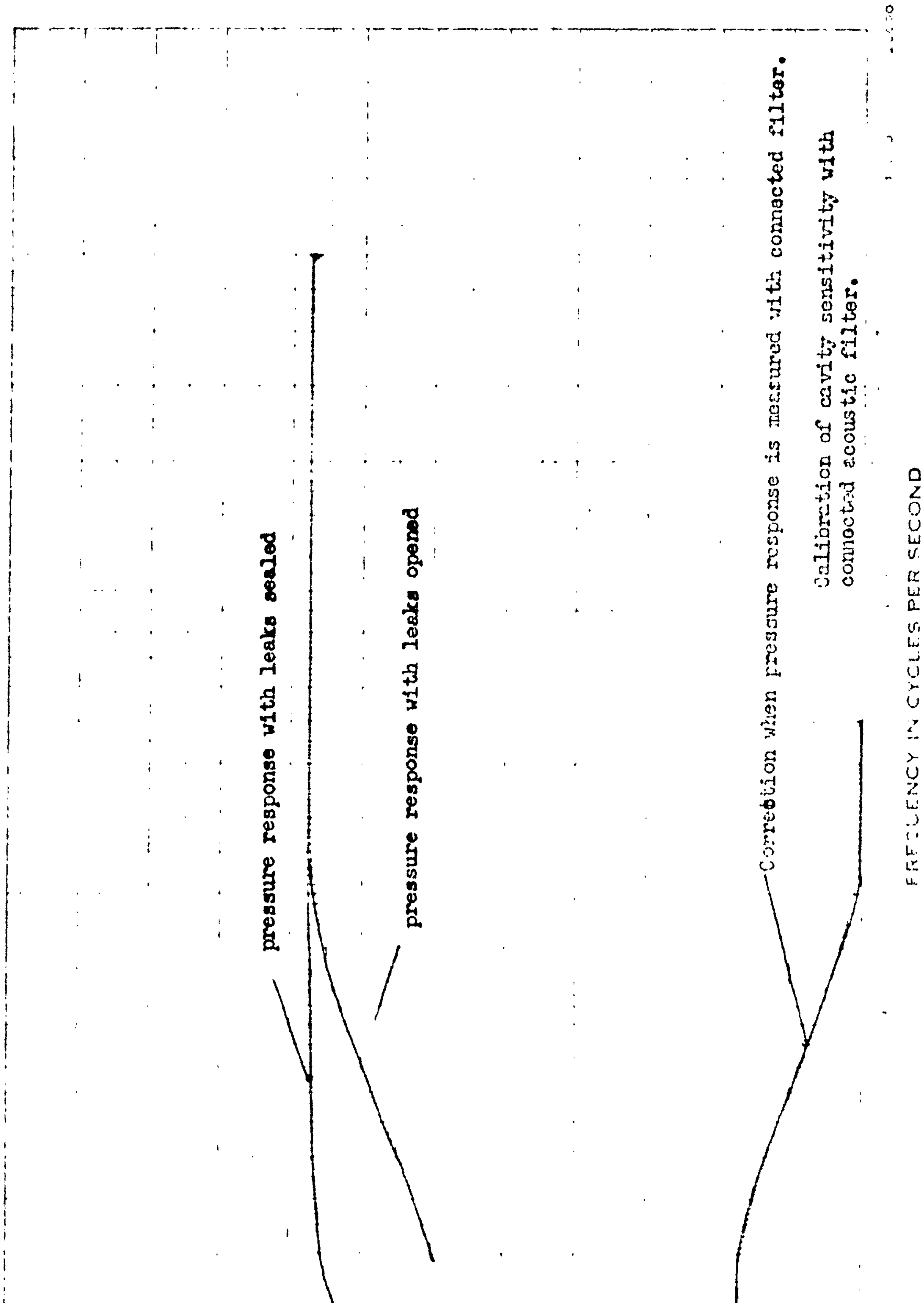
$$f = \alpha_{mn} \times \frac{c}{2r_1}$$



WESTERN ELECTRO-ACOUSTIC LABORATORY
LOS ANGELES 49, CALIFORNIA

6-28-57 F. J.

Figure 14-64



pressure response with leaks sealed
pressure response with leaks opened
Correction when pressure response is measured with connected filter.
Calibration of cavity sensitivity with connected acoustic filter.
FREQUENCY IN CYCLES PER SECOND
110
dB
SPL

Figure 14-65

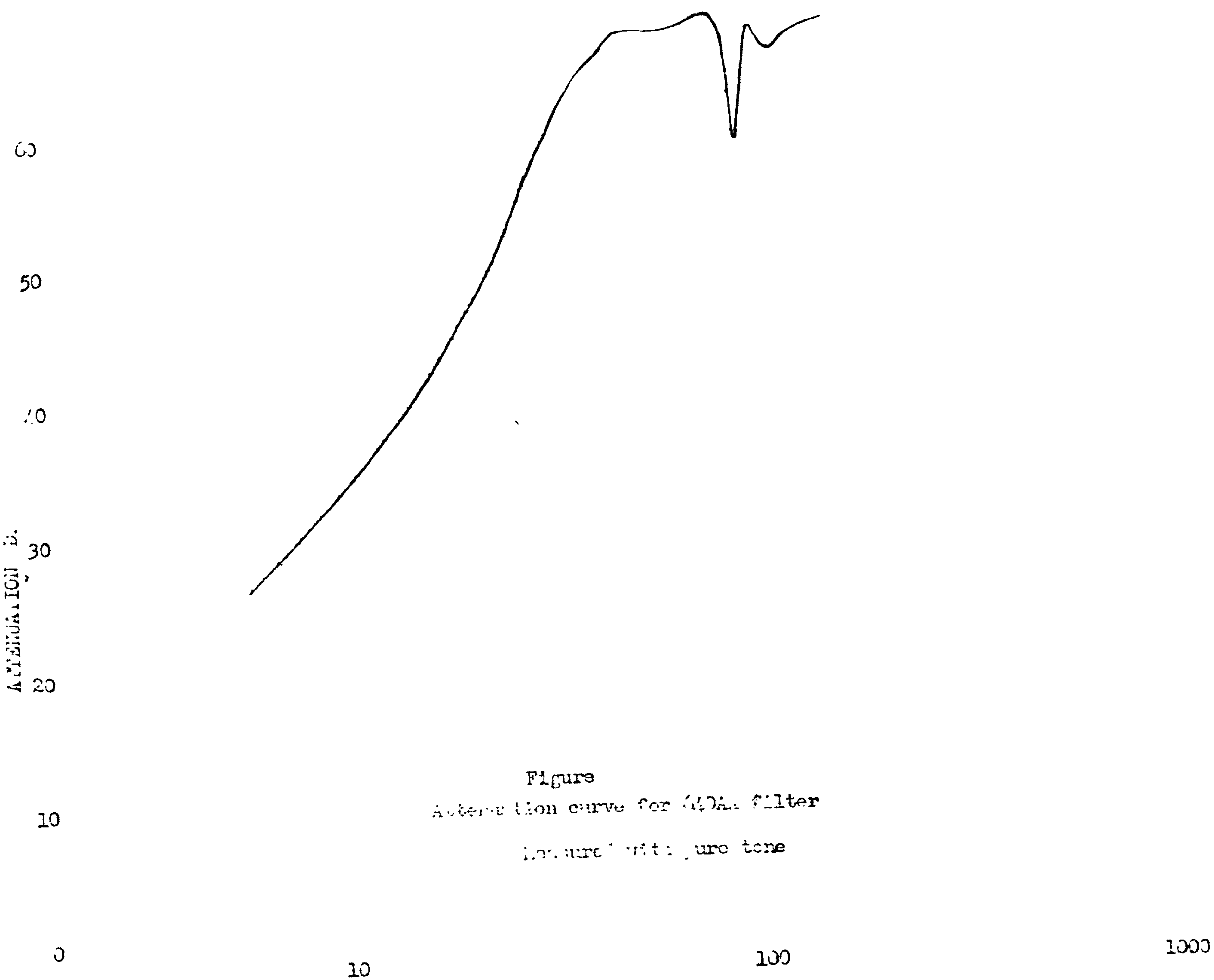


Figure
Attenuation curve for 440A filter
measured with pure tone

Figure A4-66

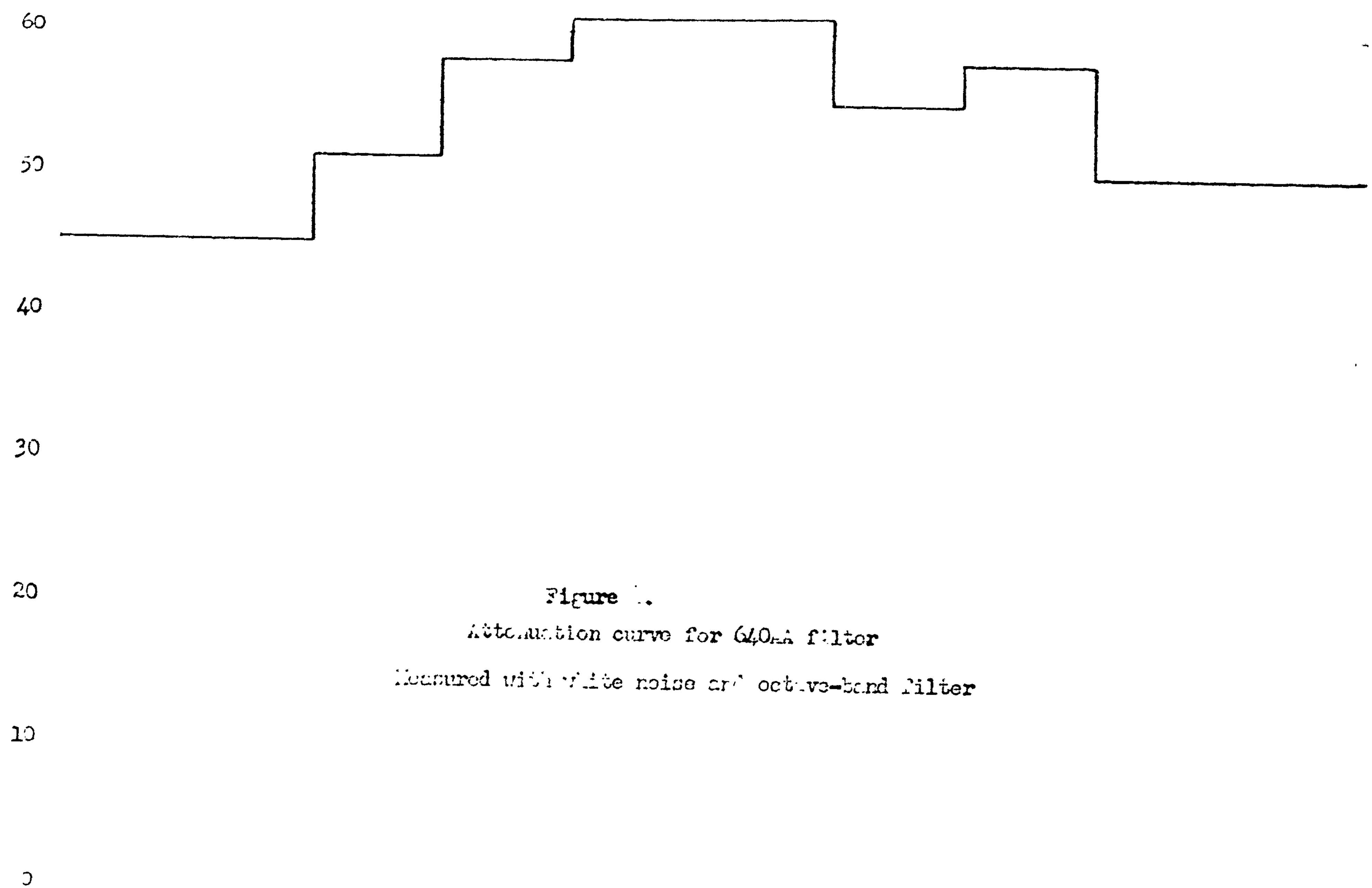
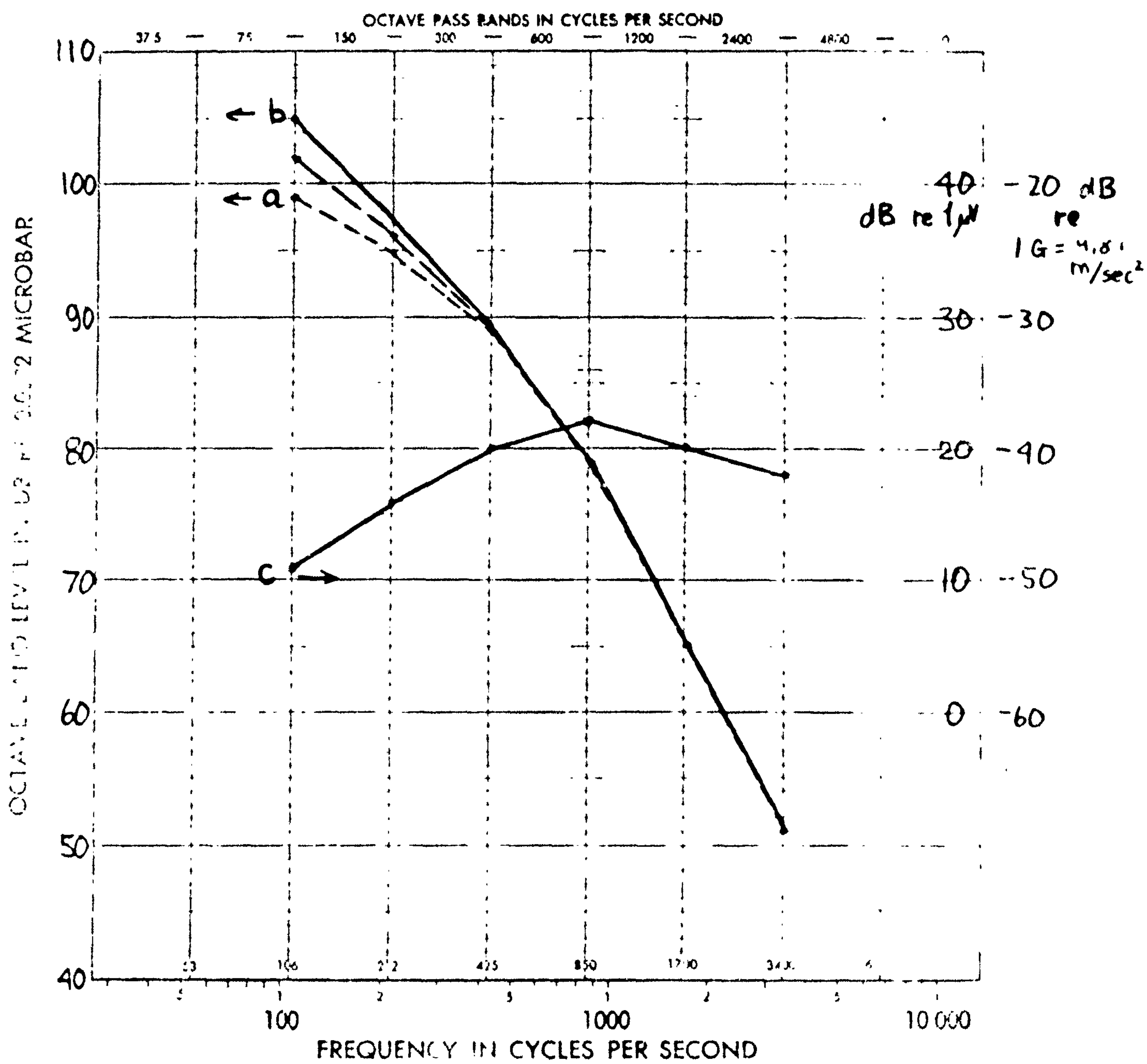


Figure .

Attenuation curve for 640 filter

Measured with white noise and octave-band filter

Figure A4-67

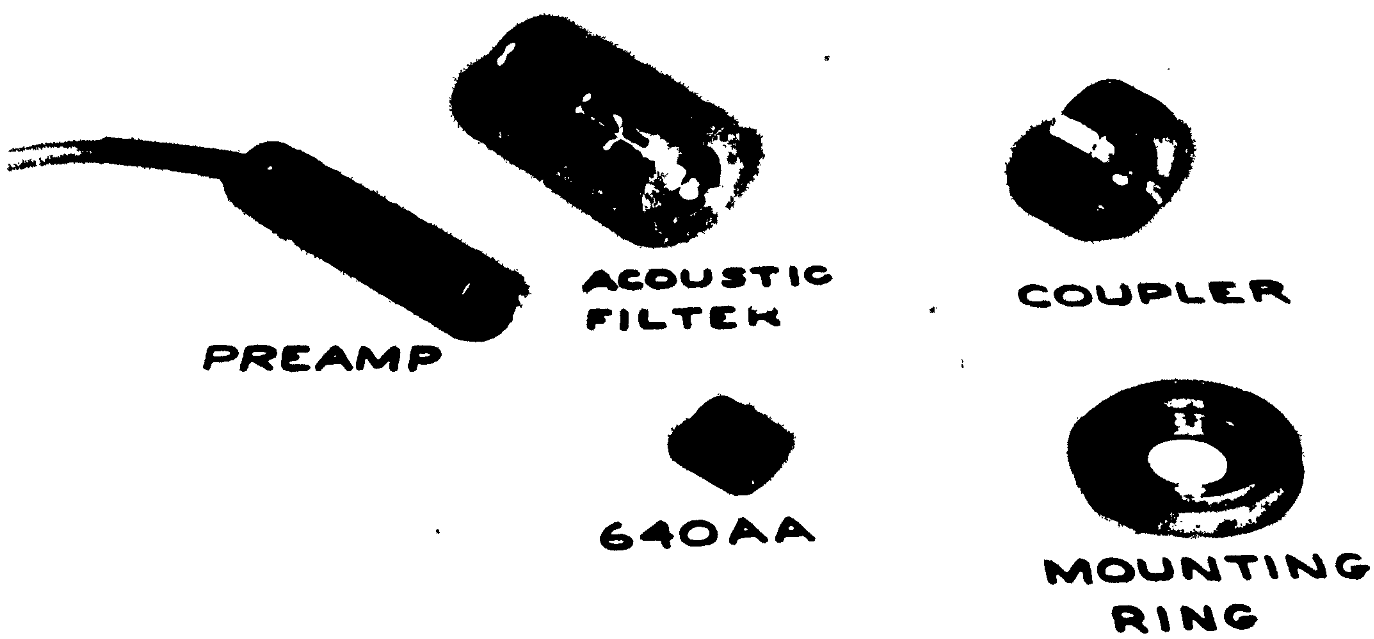


curve a : Measured pressure in the cavity in the forehead pickup

curve b : Curve a corrected with respect to leak from the cavity and resonance of the forehead pickup and the skin compliance.

curve c : Predicted output voltage from an accelerometer with sensitivity 1 mV/G.

Figure A4-68

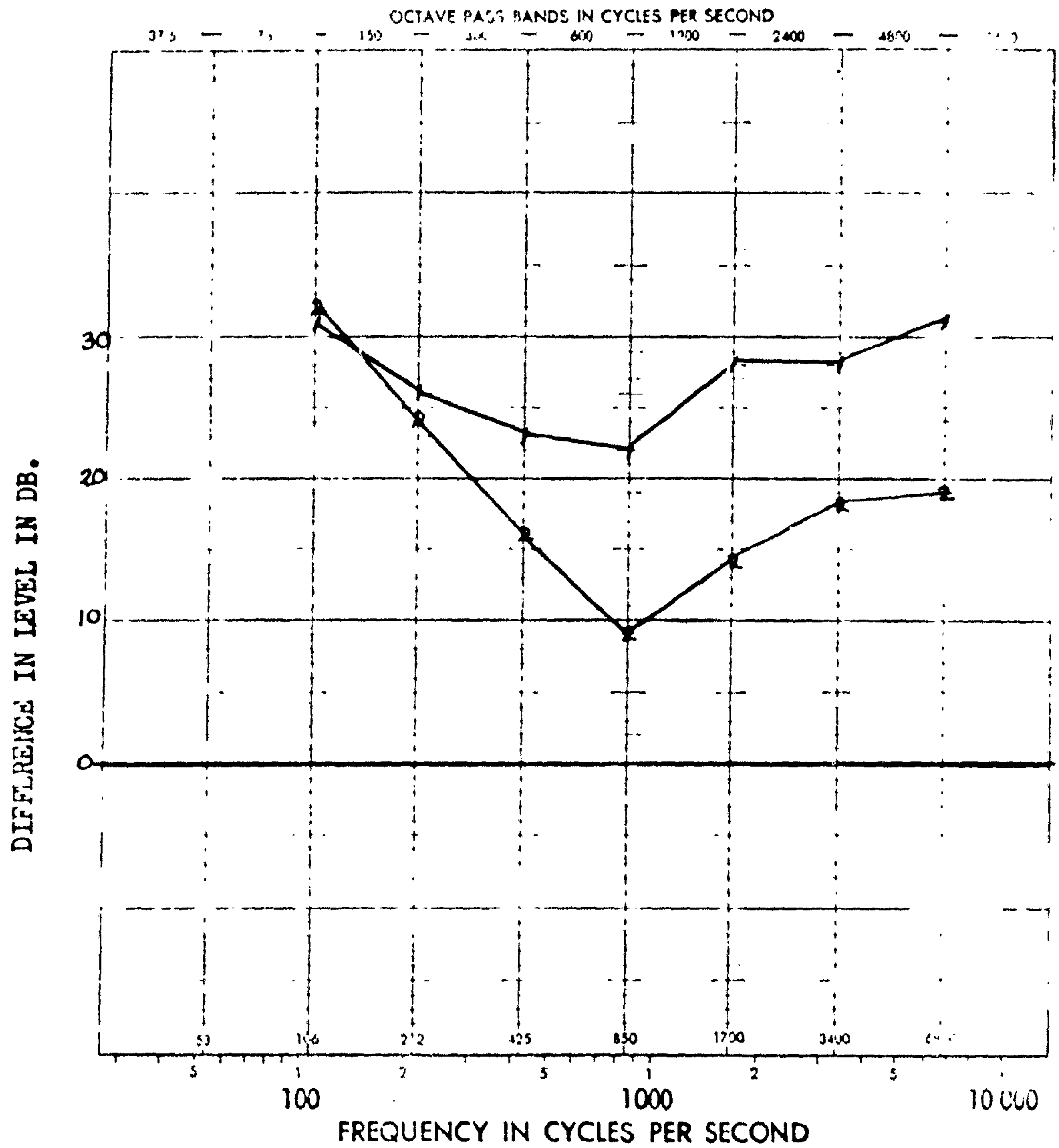


FOREHEAD
COUPLER
USING 640AA

FOREHEAD
COUPLER
USING 640AA

Figure A4 7

CALCULATED DIFFERENCE BETWEEN BONE-CONDUCTED AND AIRBORNE SPEECH ON THE FOREHEAD USING THE 640AA COUPLER PRESSED COMFORTABLY TO THE FOREHEAD.

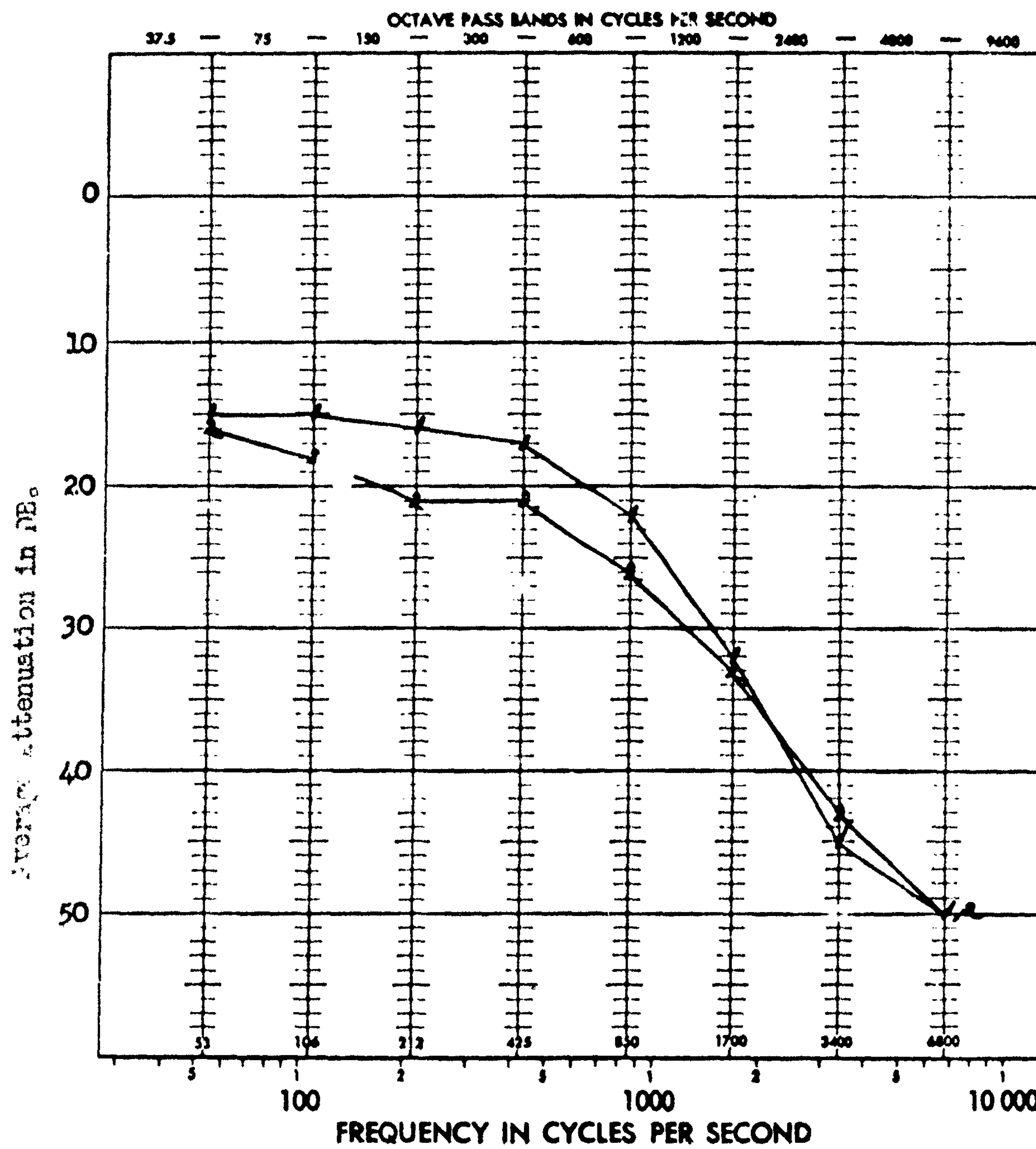


Curve 1: Separating plane between mouth and forehead

Curve 2: No separating plane

Figure A4-72

AVERAGE ATTENUATION TO AIRBORNE NOISE OF FOREHEAD PICKUP CONSISTING OF 640AA IN COUPLER WITH NO AUXILIARY DIAPHRAGM.



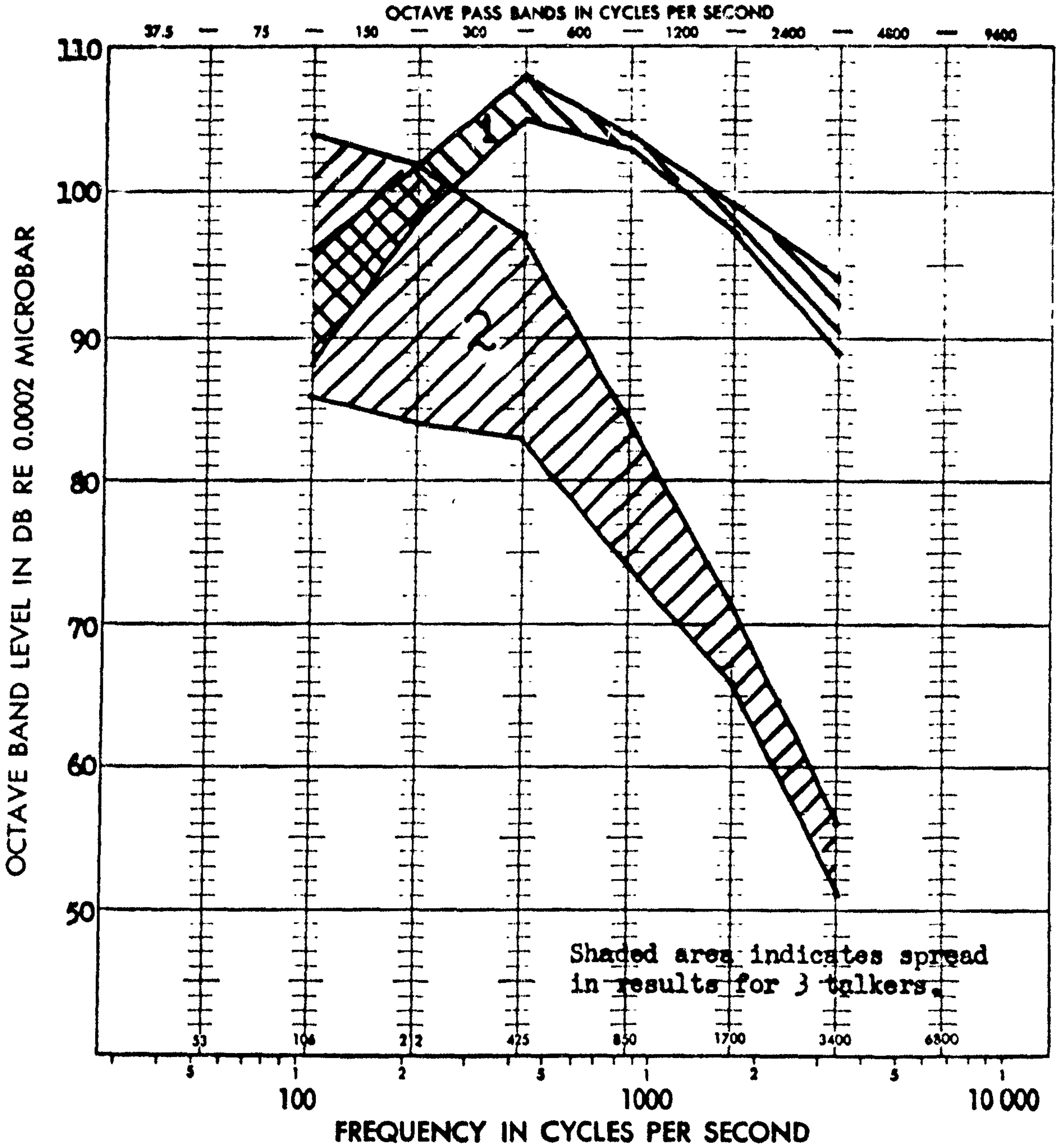
4 Runs on 4 Observers (FJ, JPC, JCC, JRS)

Curve 1: Attenuation of Airborne Sound by head P.U. pressed comfortably against forehead.

Curve 2: Attenuation of Airborne Sound by head P.U. pressed tightly against forehead.

Figure A4-73

LONG TIME AVERAGE SPEECH SPECTRUM ON THE FOREHEAD AND AT THE LIPS.



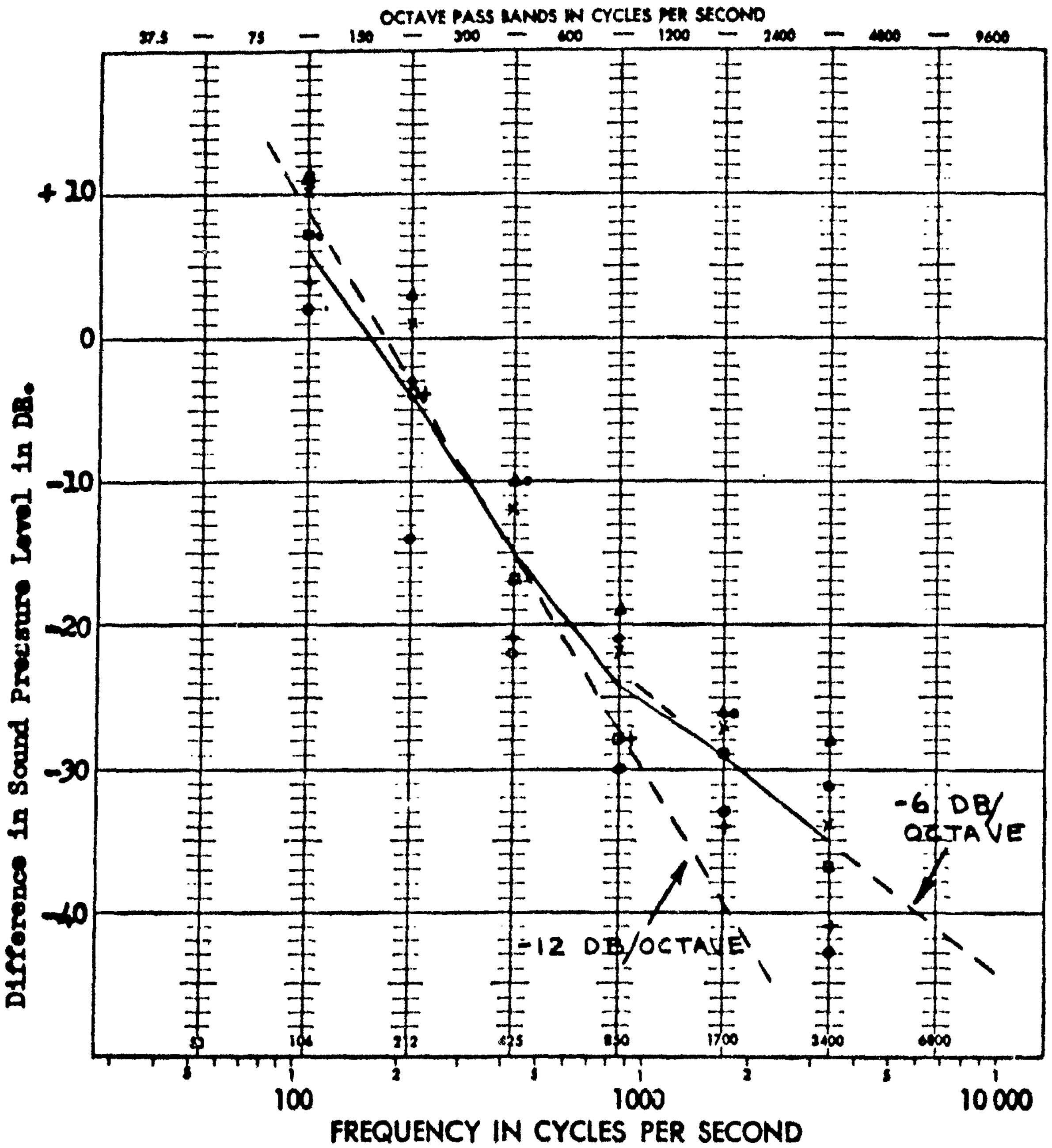
Curve 1: LTA spectrum as measured by Reference System. (Western Electric 640AA pressure probe microphone at lips). Adjusted to 110 DB over-all SPL for each talker.

Curve 2: LTA spectrum as measured by a Western Electric 640AA microphone in coupler with no auxiliary diaphragm on the forehead.

Note: Same speaking level for both conditions.

Figure A4-74

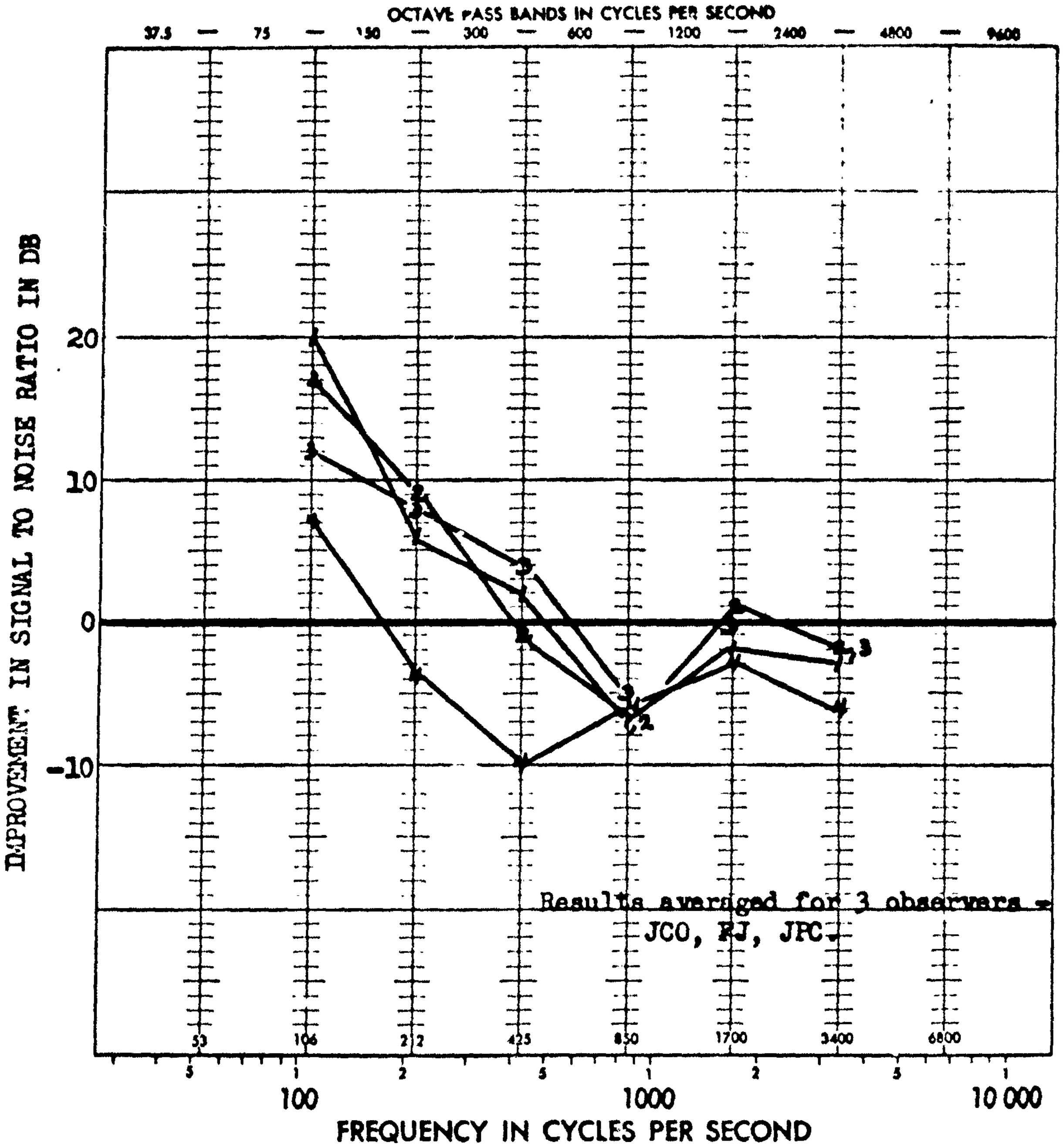
DIFFERENCE BETWEEN SOUND PRESSURE LEVEL AT LIPS AND ON FOREHEAD
WITH SAME SPEAKING LEVEL.



Note:
1. Negative values indicate level at forehead is less than at lips.
2. The points indicate results on 6 individuals.
3. The solid line is the average difference for the 6 speakers.

Figure A4-75

IMPROVEMENT IN SIGNAL TO NOISE RATIO OF SEVERAL FOREHEAD MICROPHONES RELATIVE TO THE REFERENCE SYSTEM.



Curve 1: W.E. 640AA mic. in forehead coupler. No auxiliary diaphragm.
Curve 2: W.E. 640AA mic. in forehead coupler. Auxiliary aluminum diaphragm .025" thick.
Curve 3: Magnetic mic. with diaphragm in contact with forehead.
Curve 4: Magnetic mic. with water pillows between diaphragm and forehead.

Reference System is a W.E. 640AA pressure probe microphone at the lips in the open.

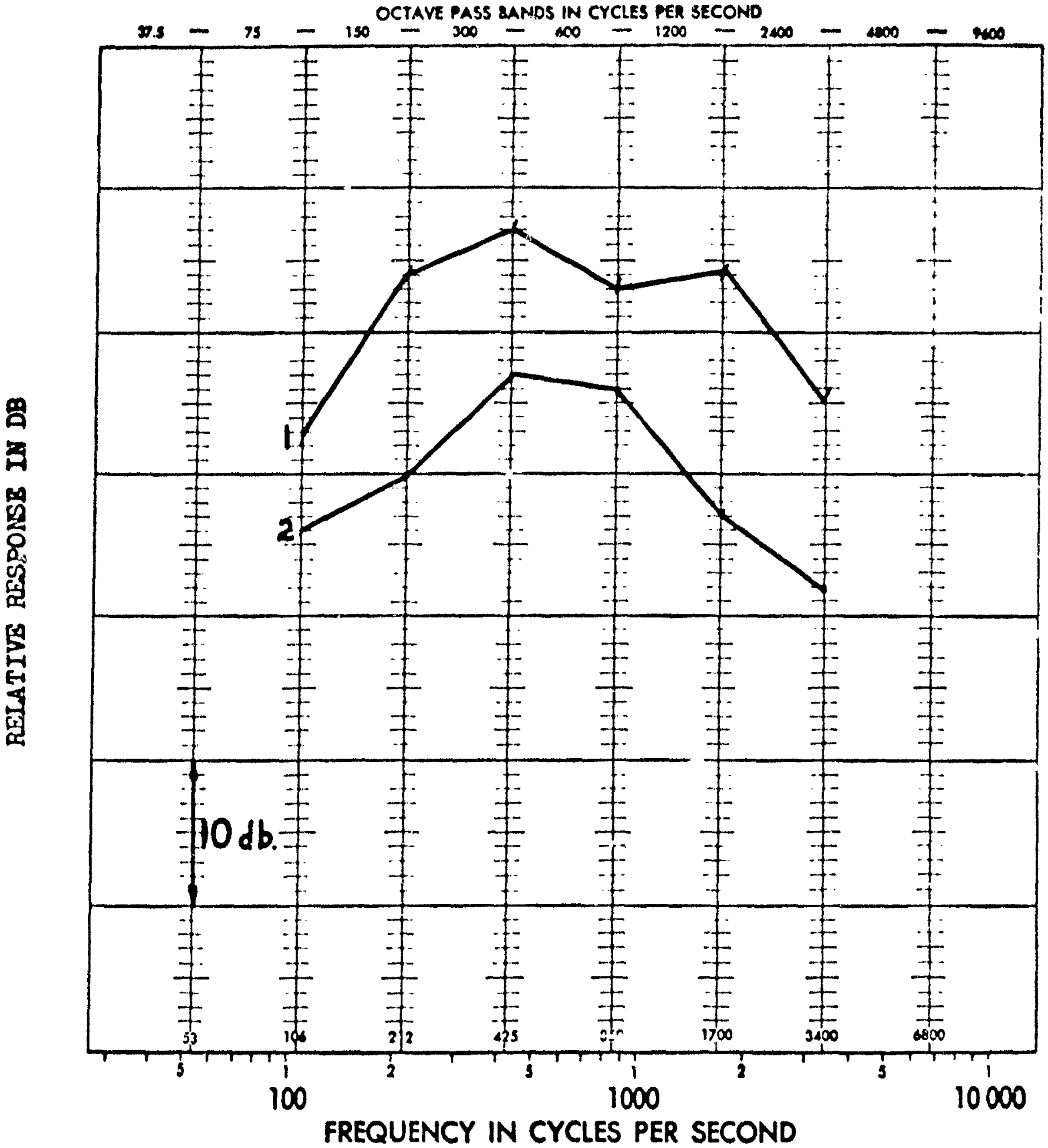
Figure A4-76

MAGNETIC
FOREHEAD
PICKUP


MAGNETIC
FOREHEAD
PICKUP

Figure A-14

LONG TIME AVERAGE SPEECH SPECTRA USING MAGNETIC FOREHEAD MICROPHONE WITH DIAPHRAGM DIRECTLY AGAINST THE HEAD.



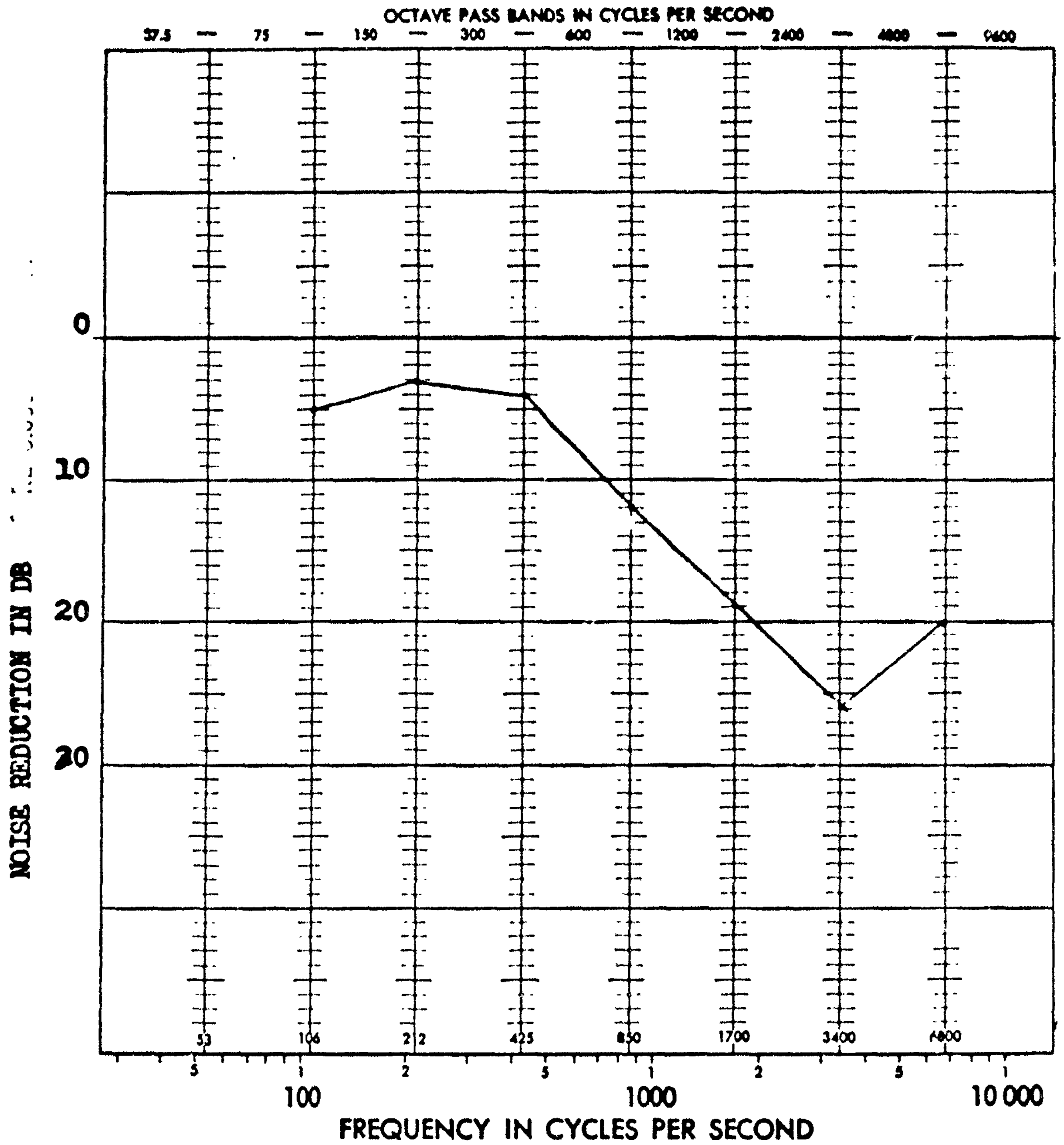
Curve 1: Equalized forehead microphone.

Curve 2: Reference System (W.E. 640AA pressure probe microphone at the lips in the open).

Note: Approximately "normal conversational level" for all spectra.

Figure A4-80

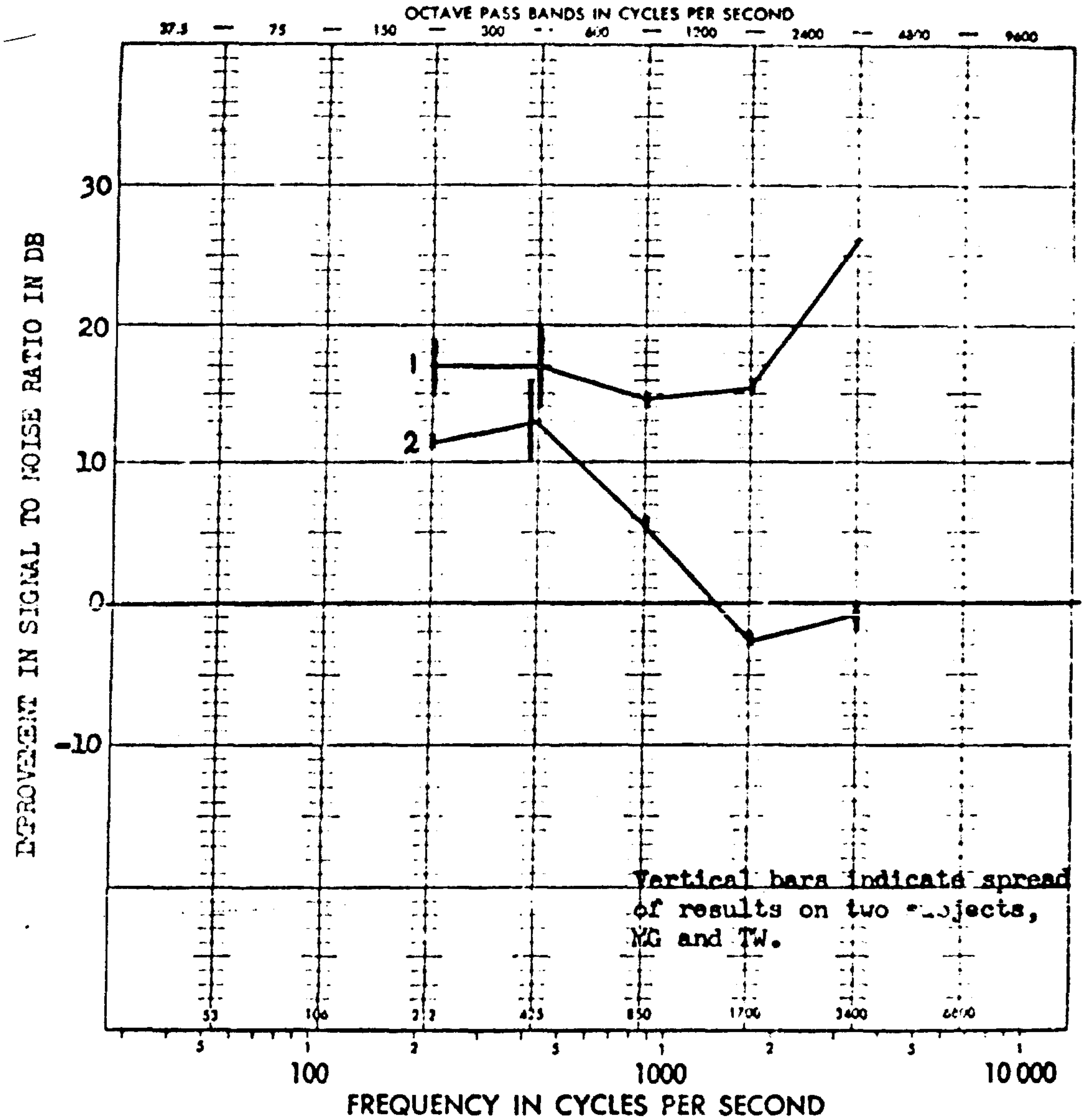
NOISE REDUCTION GAINED WHEN MA-1 helmet
IS WORN OVER FORHEAD MICROPHONE.



Average of 4 runs on 2 observers.

Figure A4-81

IMPROVEMENT IN SIGNAL TO NOISE RATIO OF MAGNETIC FOREHEAD MICROPHONE RELATIVE TO THE REFERENCE SYSTEM.



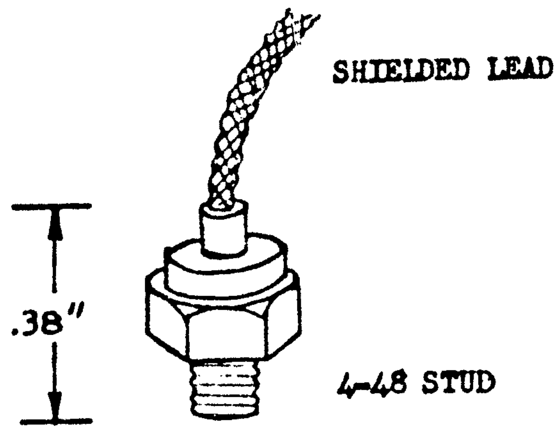
Curve 1: Magnetic forehead microphone under MA-1 Helmet.

Curve 2: Magnetic forehead microphone with no additional noise shielding.

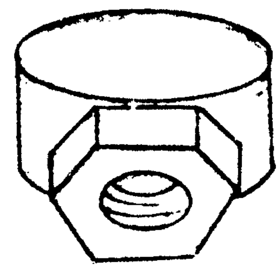
Note: Diaphragm placed directly on forehead.
Reference system is a W.E. 640AA pressure probe microphone at the lips in the open.

Figure A4-82

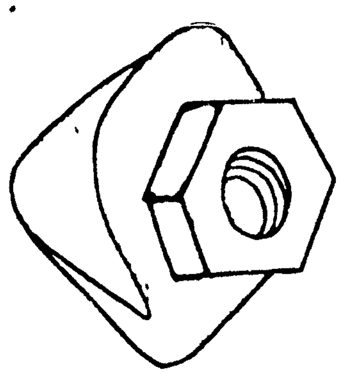
SPEECH PICKUP ON THE TEETH



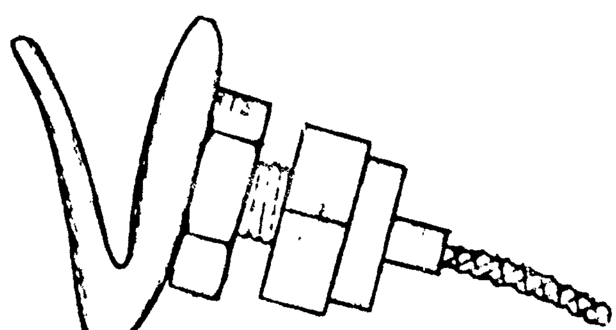
COLUMBIA RESEARCH
ACCELEROMETER MODEL 607



BAND TO HOLD
ACCELEROMETER TO
TOOTH

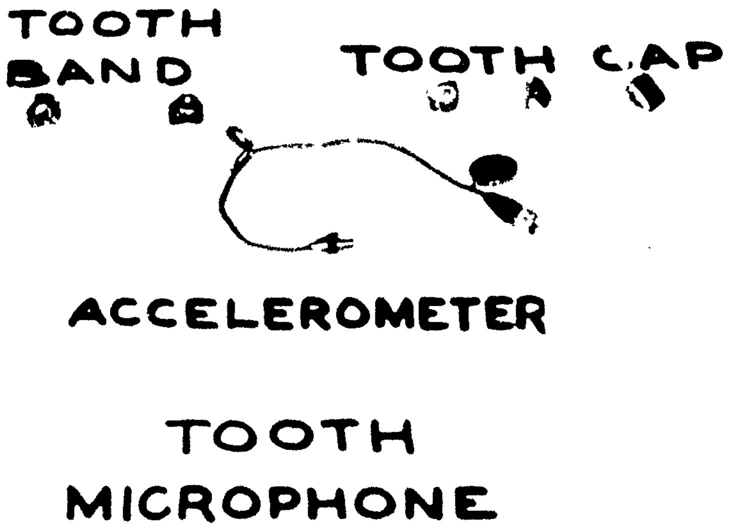


TOOTH CAP



ACCELEROMETER
MOUNTED ON TOOTH CAP

Figure A4-83



8 1 2 A4-84

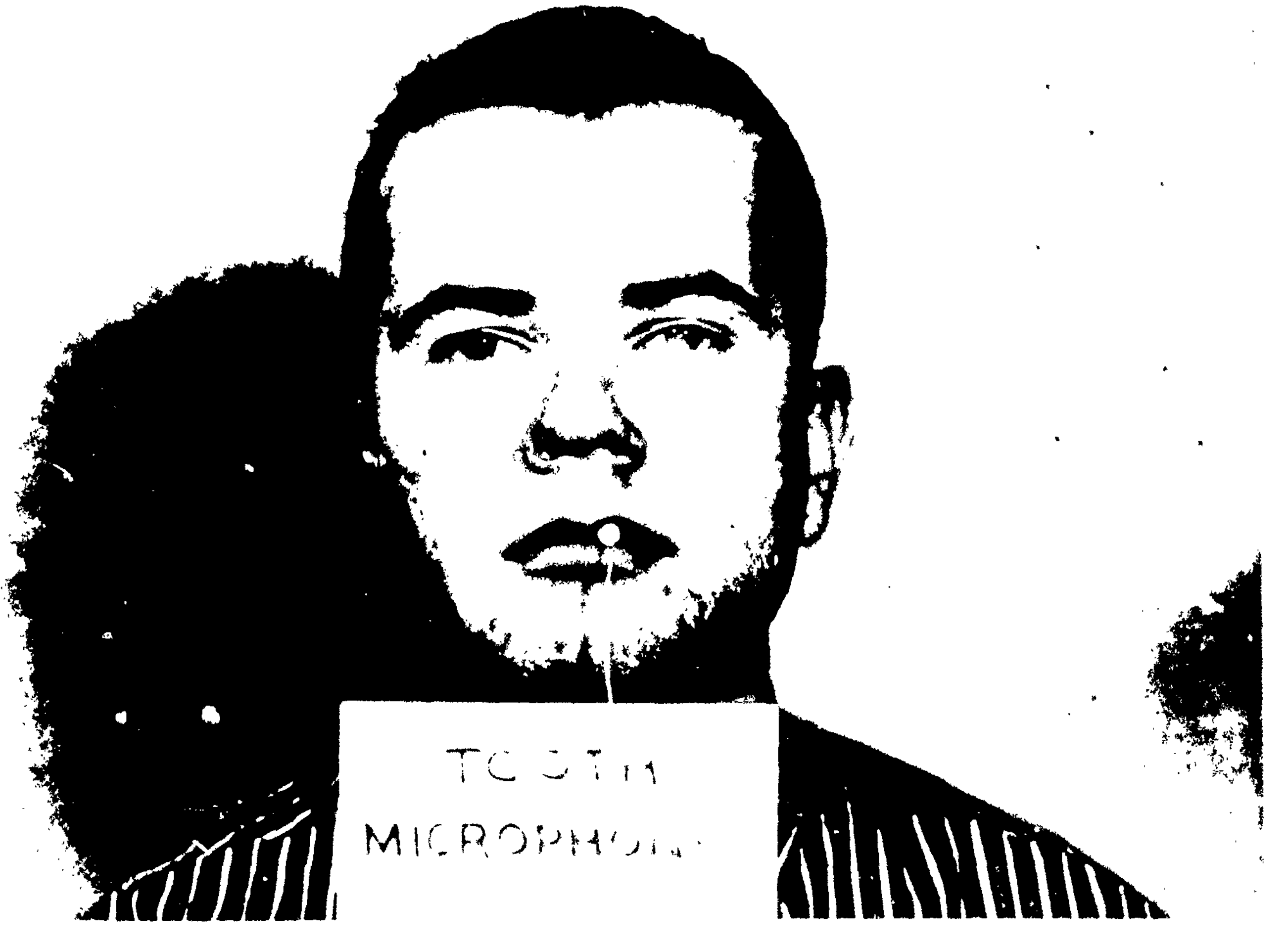
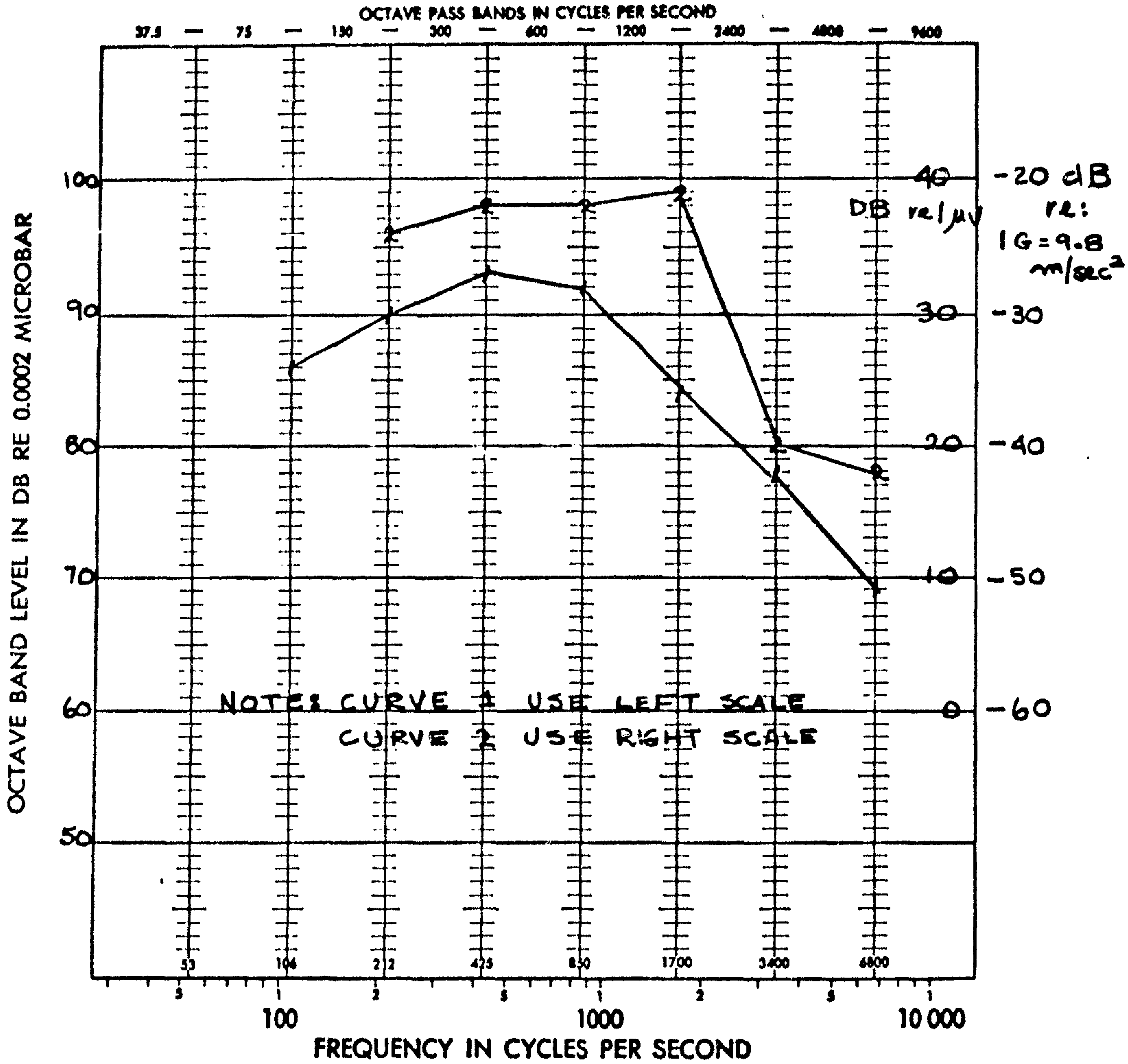


Figure A4 85

Figure A4.56

ABSOLUTE AMPLITUDE OF THE SPEECH SPECTRUM ON THE LEFT UPPER CENTRAL INCISOR. SPEAKER: TW.

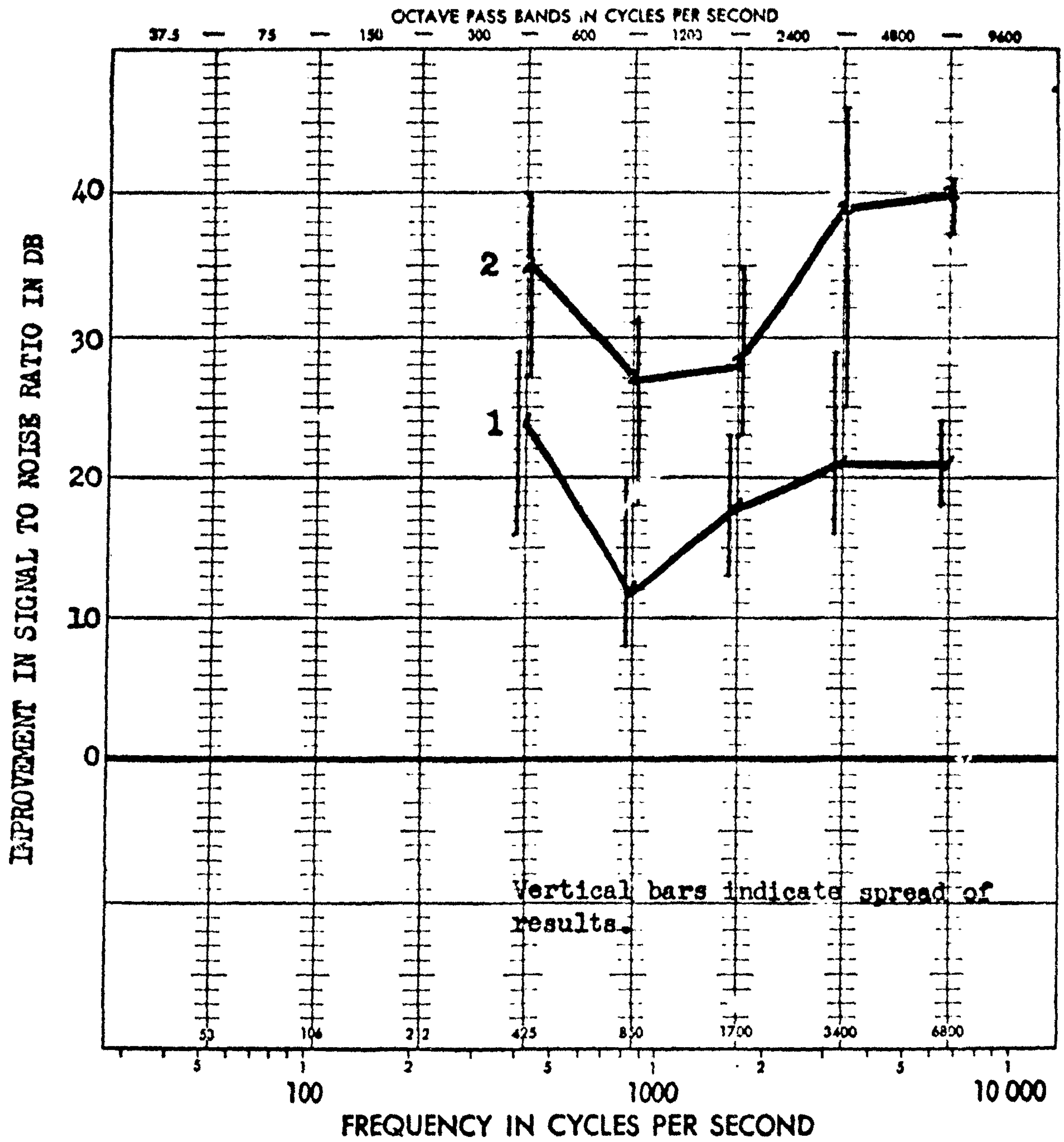


Curve 1: Sound pressure level at the lips.
Curve 2: Speech spectrum on the left upper central incisor.

Note: Same speaking effort for both measurements.

Figure A4-88

IMPROVEMENT IN SIGNAL TO NOISE RATIO OF TOOTH
MICROPHONE RELATIVE TO THE REFERENCE SYSTEM



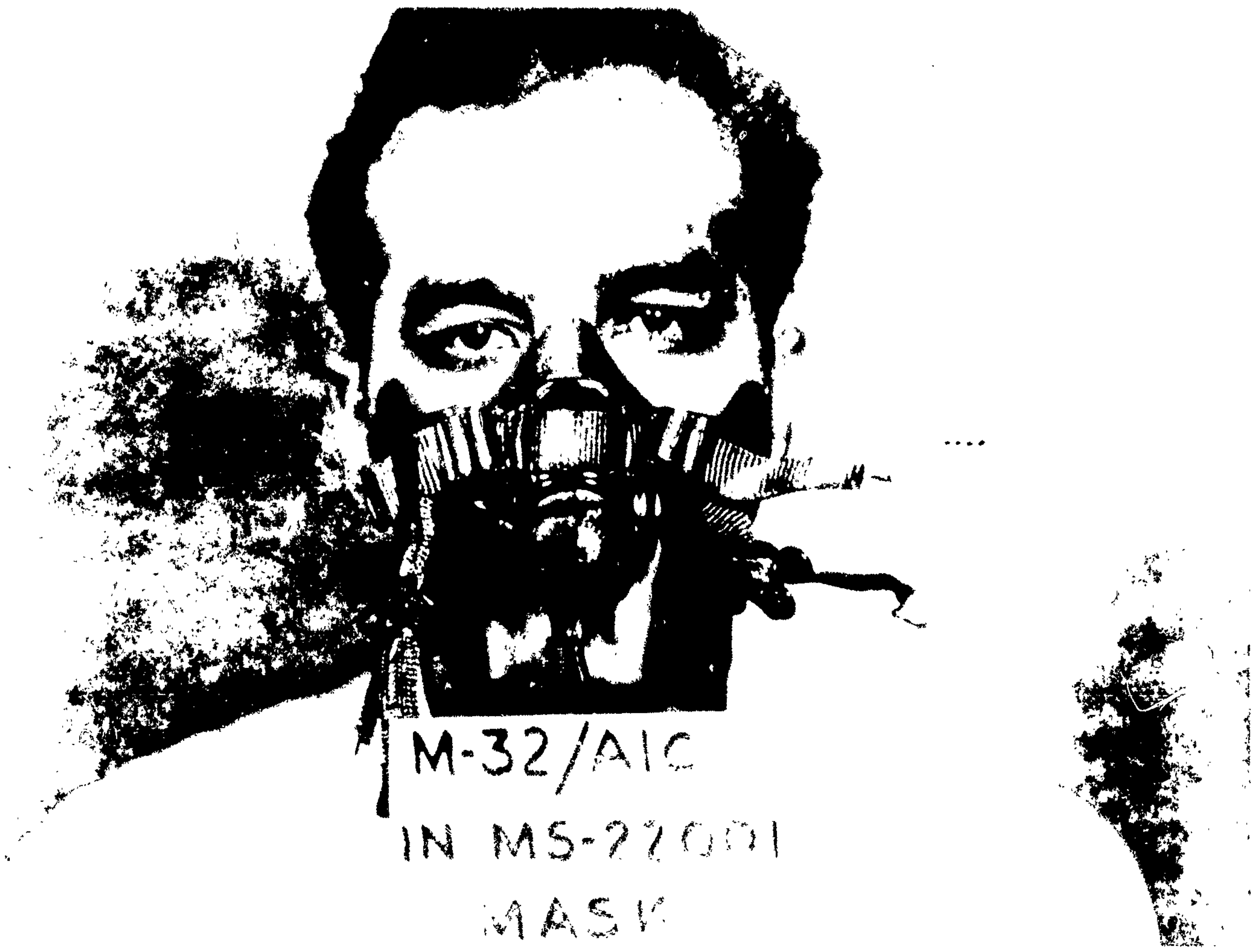
Curve 1: Tooth microphone in open.
Curve 2: Tooth microphone in MA-1 Helmet.

1. Results are averaged for observers TW, MG and JPC.

2. Reference System is Western Electric 640AA pressure probe microphone at the lips in the open.

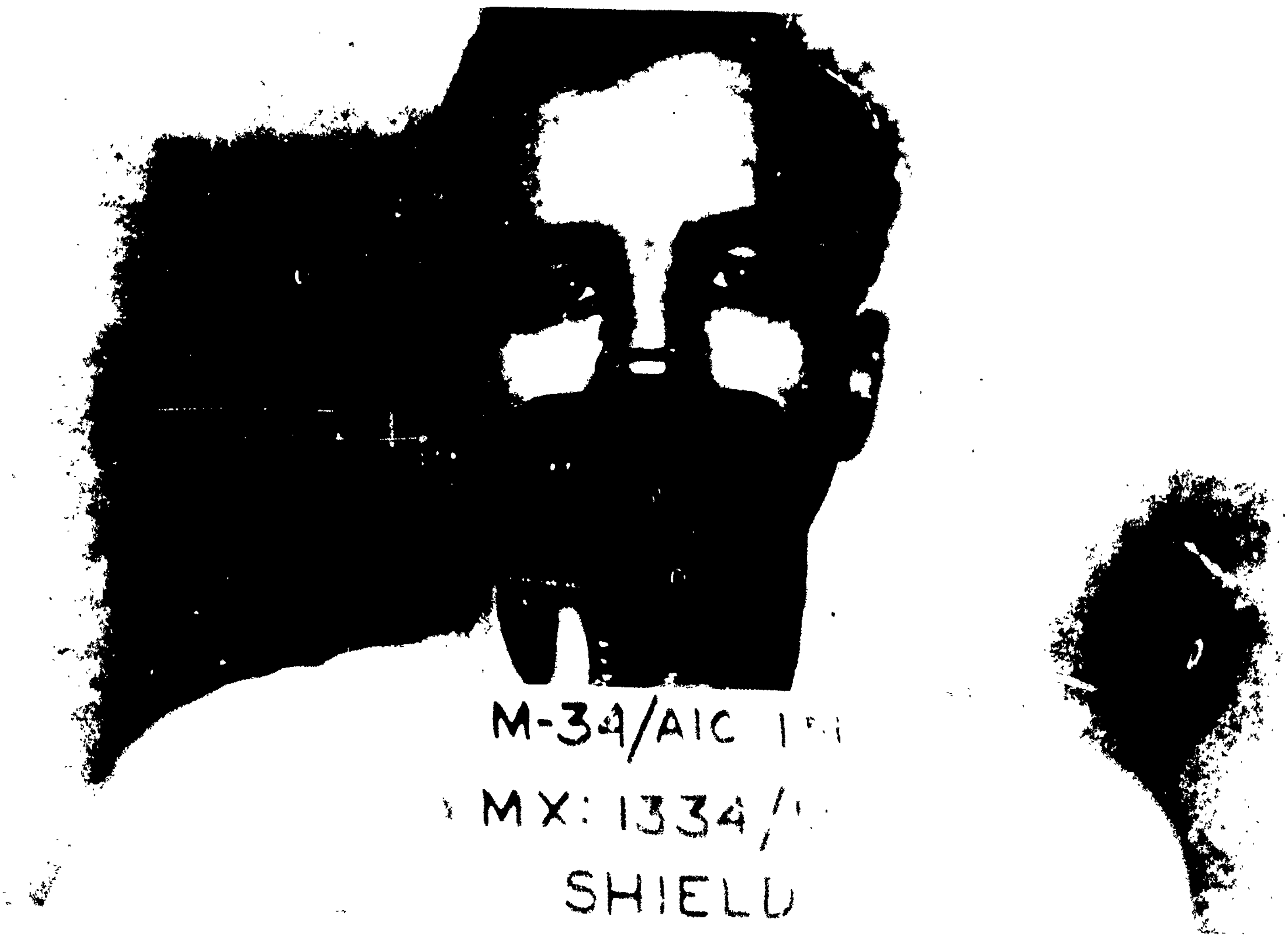
Figure A4-89


M-33/AIC
M-34/AIC IN
MX: 1334/U
SHIELD
M-32/AIC
IN MS-22001
MASK

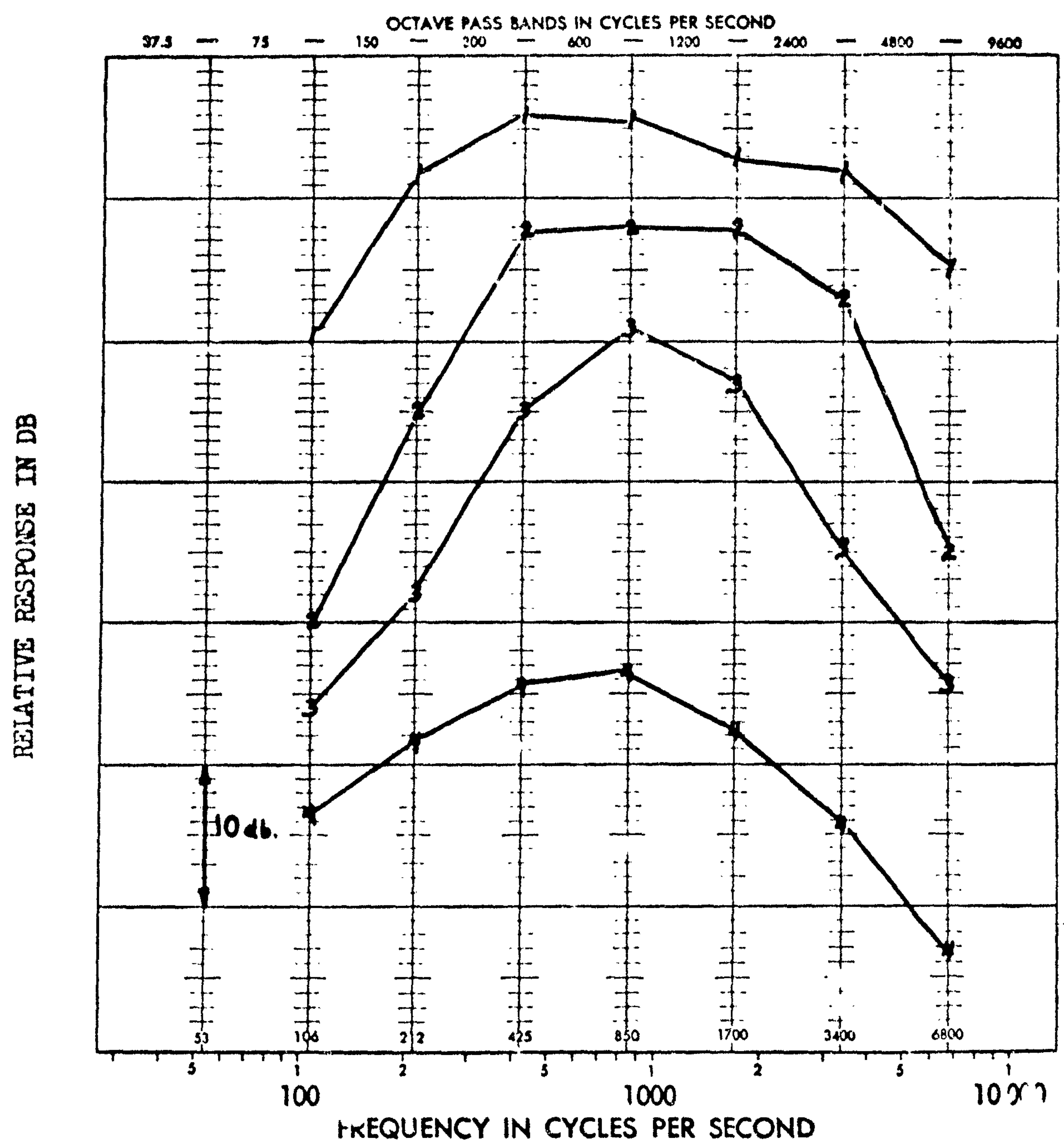

M-32/AIC
IN MS-22001
MASK

M-33/AI

Figure A5.9.


M-34/AIC
MX: 1334/
SHIEL

LONG TIME AVERAGE SPEECH SPECTRA FOR M-32, M-33, AND M-34 MICROPHONES. SPEAKER: JPC.



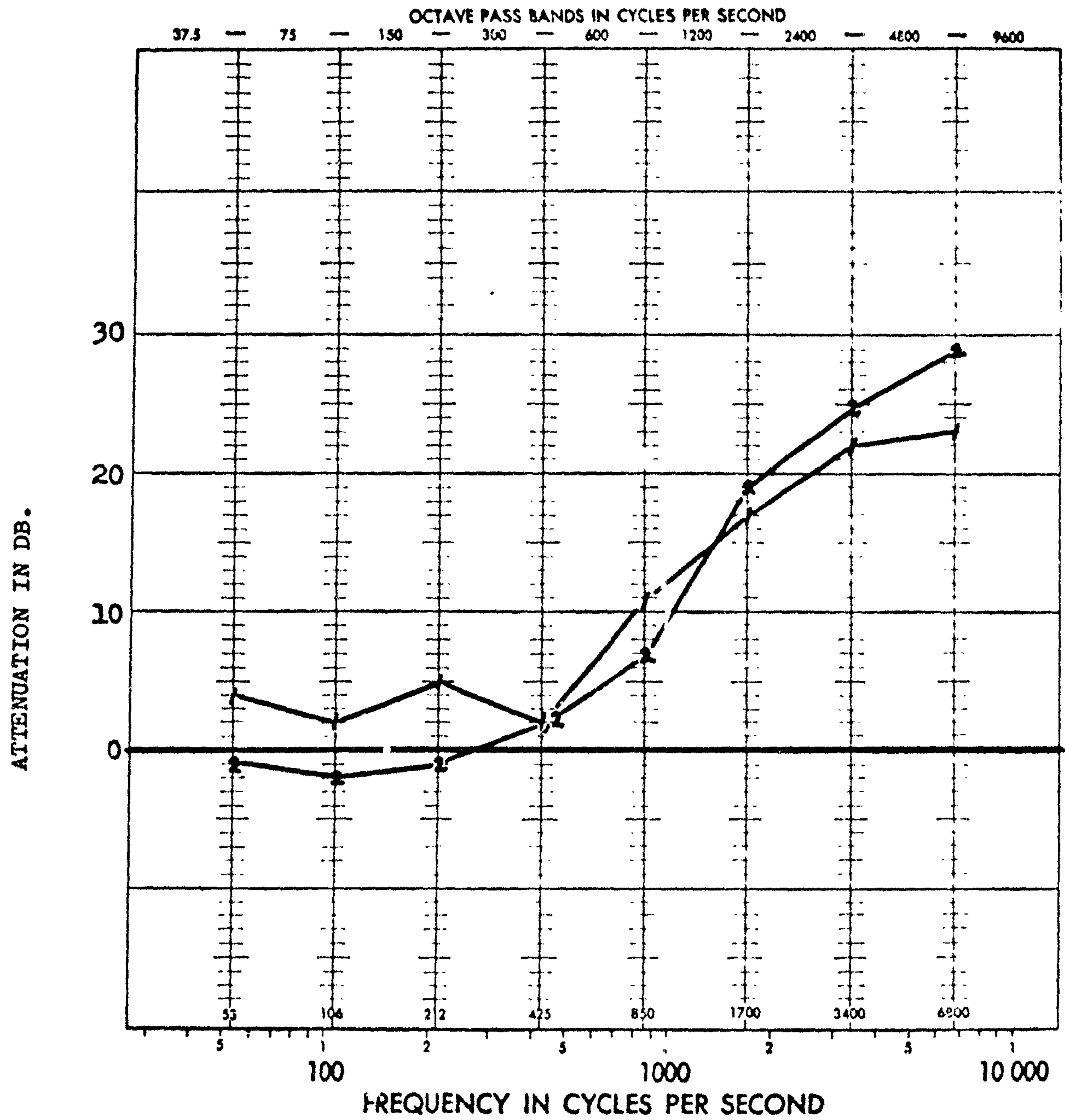
Curve 1: M-32/AIC mic. in MS 22001 Oxygen Mask.
Curve 2: M-33/AIC mic. in open.
Curve 3: M-34/AIC mic. in MX: 1334/U noise shield.
Curve 4: Reference System (Western Electric 640AA pressure probe microphone at the lips in the open).

Note: Approximately "Normal Conversational Level" for all spectra.

Figure A4-94

ATTENUATION OF OXYGEN MASK AND NOISE SHIELD USED WITH

AIC-10 MICROPHONES



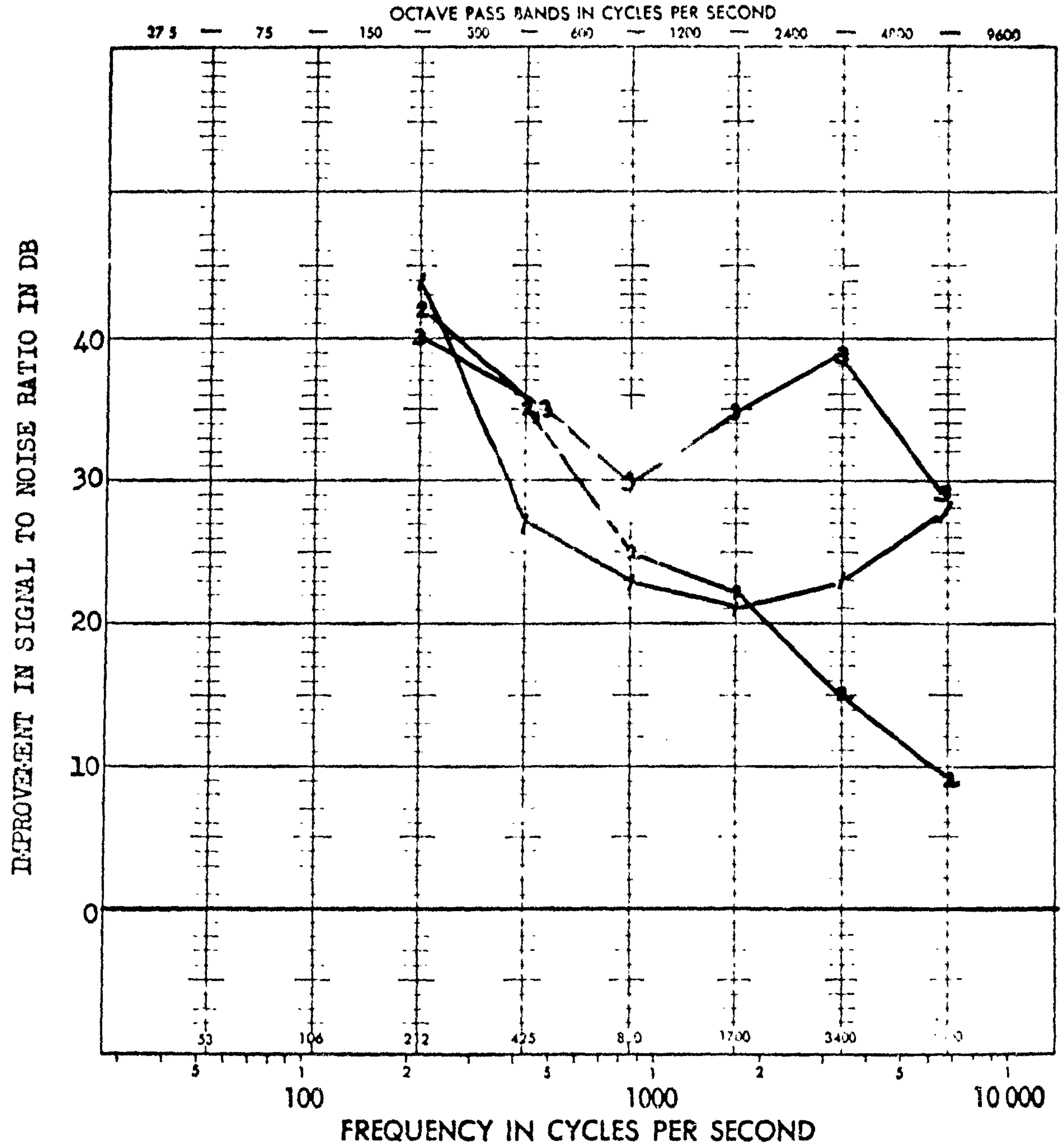
Curve 1: Attenuation of MS 22001 Oxygen Mask

Curve 2: Attenuation of MX:1334/U Noise Shield

Average of 2 runs on each device on 2 observers.

Figure A4-95

IMPROVEMENT IN SIGNAL TO NOISE RATIO OF M-32/AIC, M-33/AIC, and M-34/AIC RELATIVE TO THE REFERENCE SYSTEM. SPEAKER: JPC.

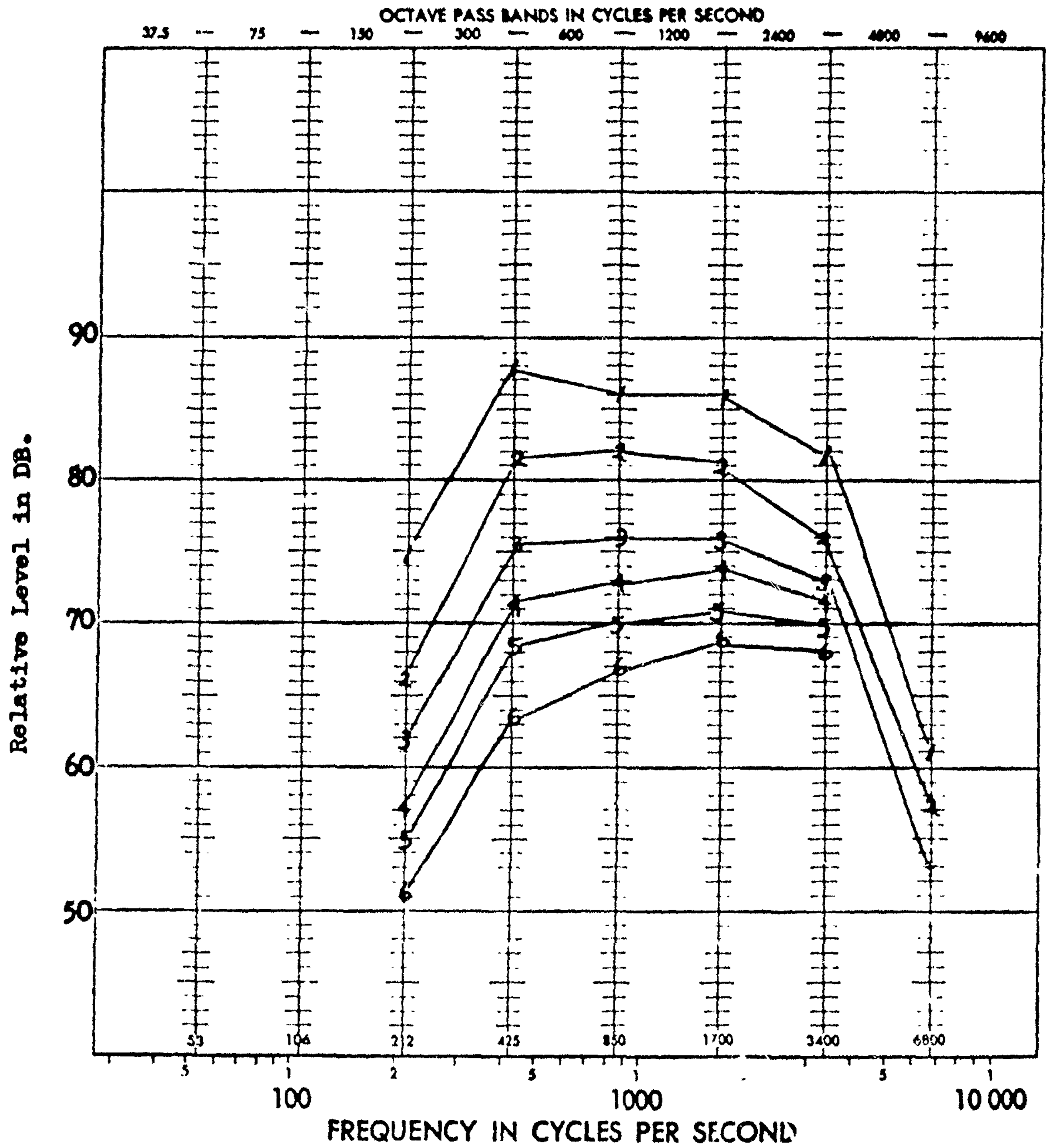


Curve 1: M-32 in MS 22001 Oxygen Mask.
Curve 2: M-33 in open.
Curve 3: M-34 in MX: 1334/U noise shield.

Reference System is a W.E. 640A pressure probe microphone at the lips in the open.

Figure A4-96

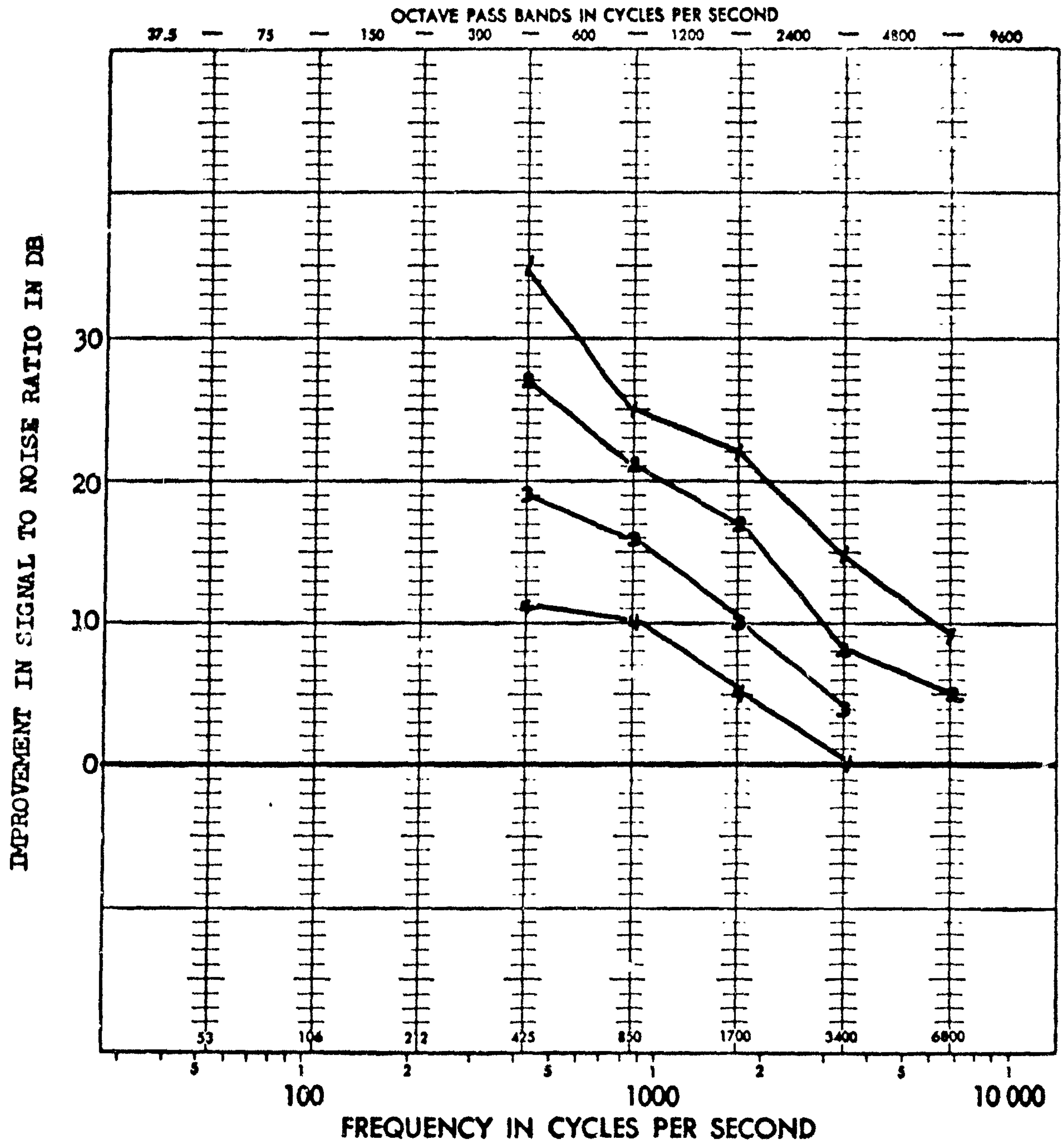
RESPONSE OF M33/AIC MICROPHONE FOR CONSTANT SPEECH LEVEL VS DISTANCE FROM LIPS



Curve 1: Microphone pressed against lips.
Curve 2: Microphone 1/8" from lips.
Curve 3: Microphone 1/4" from lips.
Curve 4: Microphone 1/2" from lips.
Curve 5: Microphone 3/4" from lips.
Curve 6: Microphone 1" from lips.

Figure A4-97

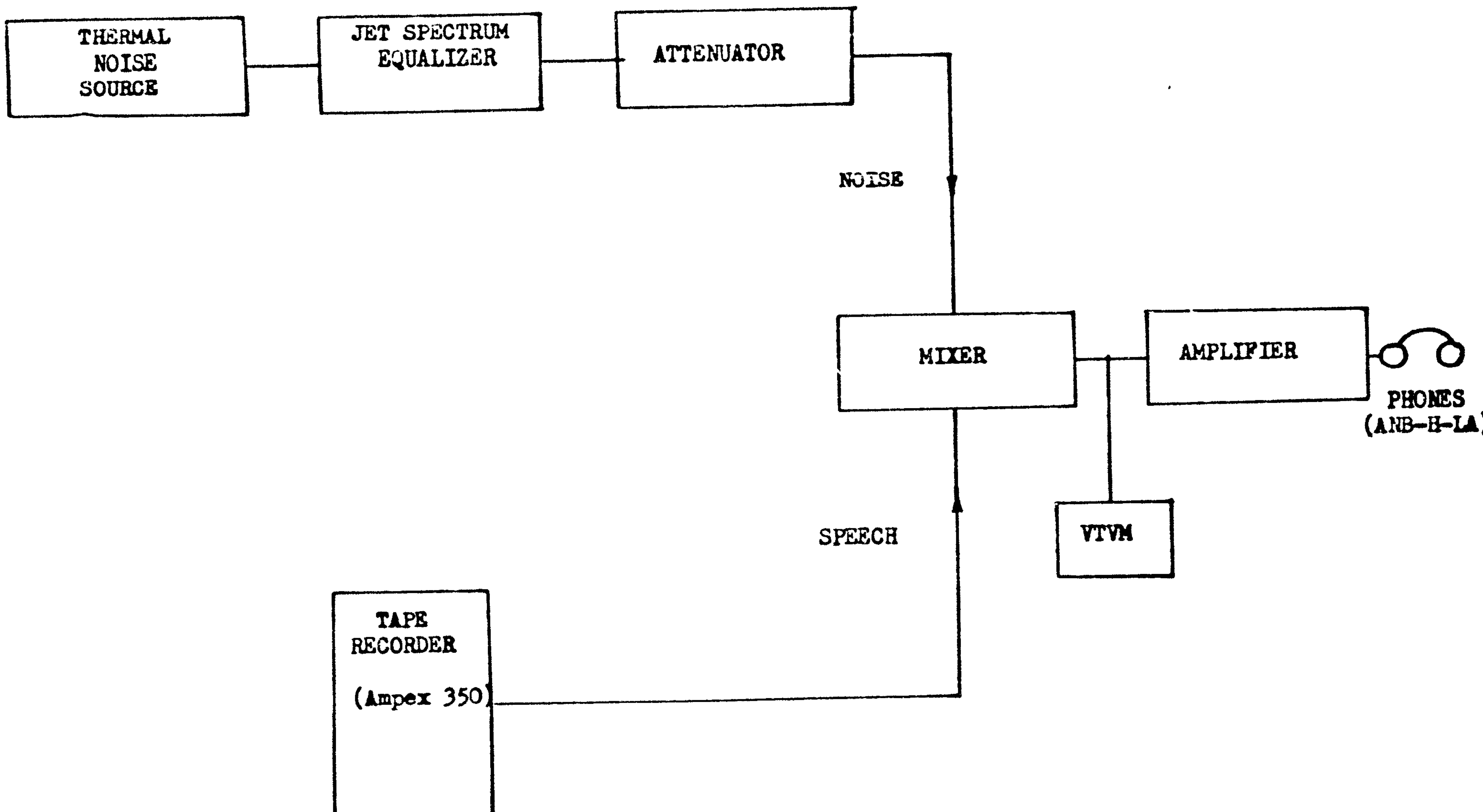
EFFECT ON THE IMPROVEMENT IN SIGNAL TO NOISE RATIO OF THE M-33/AIC MICROPHONE RELATIVE TO THE REFERENCE SYSTEM AS BOTH ARE MOVED AWAY FROM THE LIPS.



Curve 1: Both microphones at lips.
Curve 2: Both microphones 1/8" from the lips.
Curve 3: Both microphones 1/2" from the lips.
Curve 4: Both microphones 1" from the lips.

Reference System is W.E. 540AA pressure probe microphone at the lips in the open.

Figure A4-98



APPARATUS FOR MEASURING ARTICULATION SCORE VS. SIGNAL-TO-NOISE RATIO

Figure A4 - 99

CONSONANT ARTICULATION SCORE vs. SIGNAL TO NOISE RATIO FOR SPECIAL CVC WORDS USED IN ARTICULATION TESTING

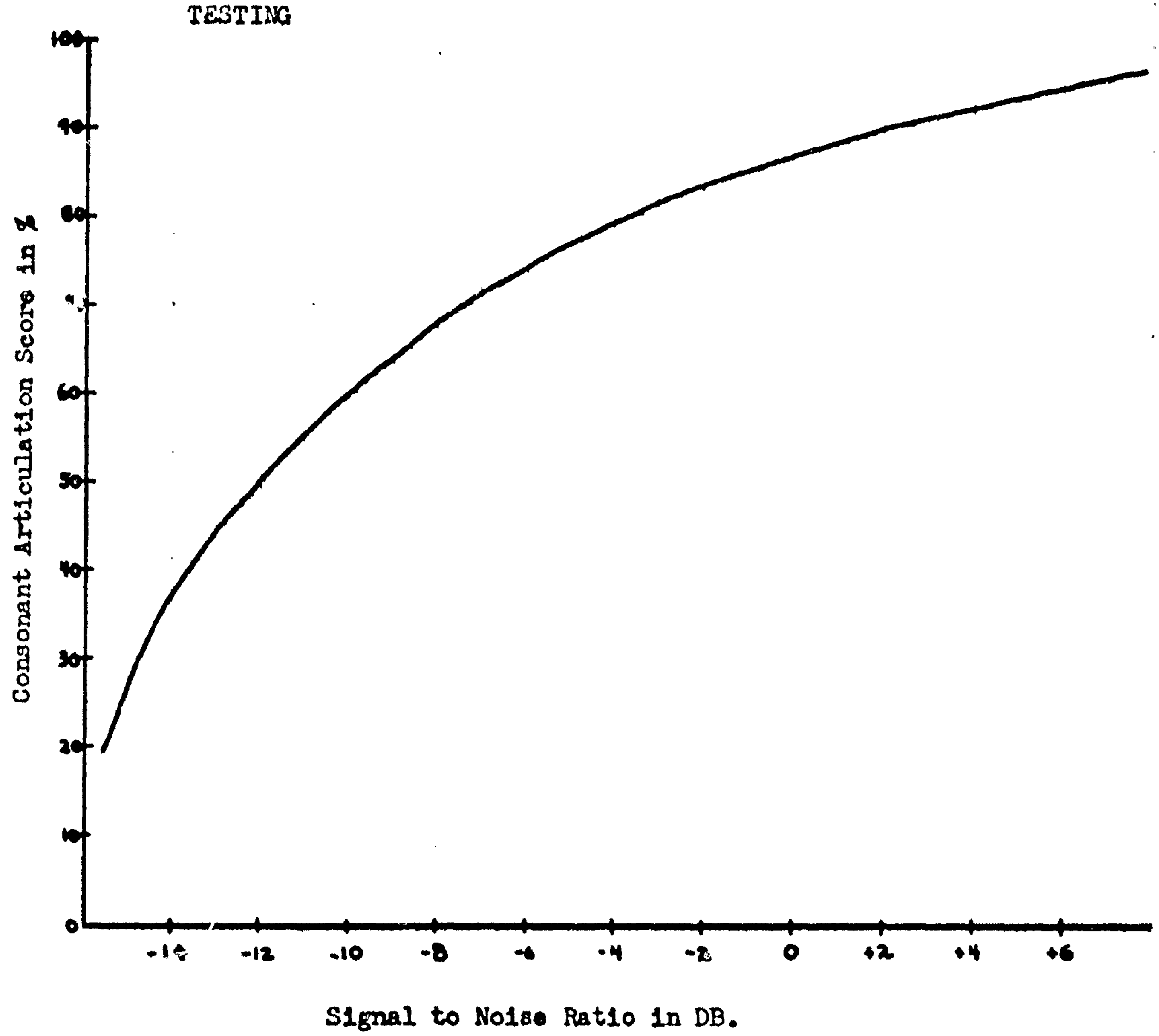


Figure A4-100

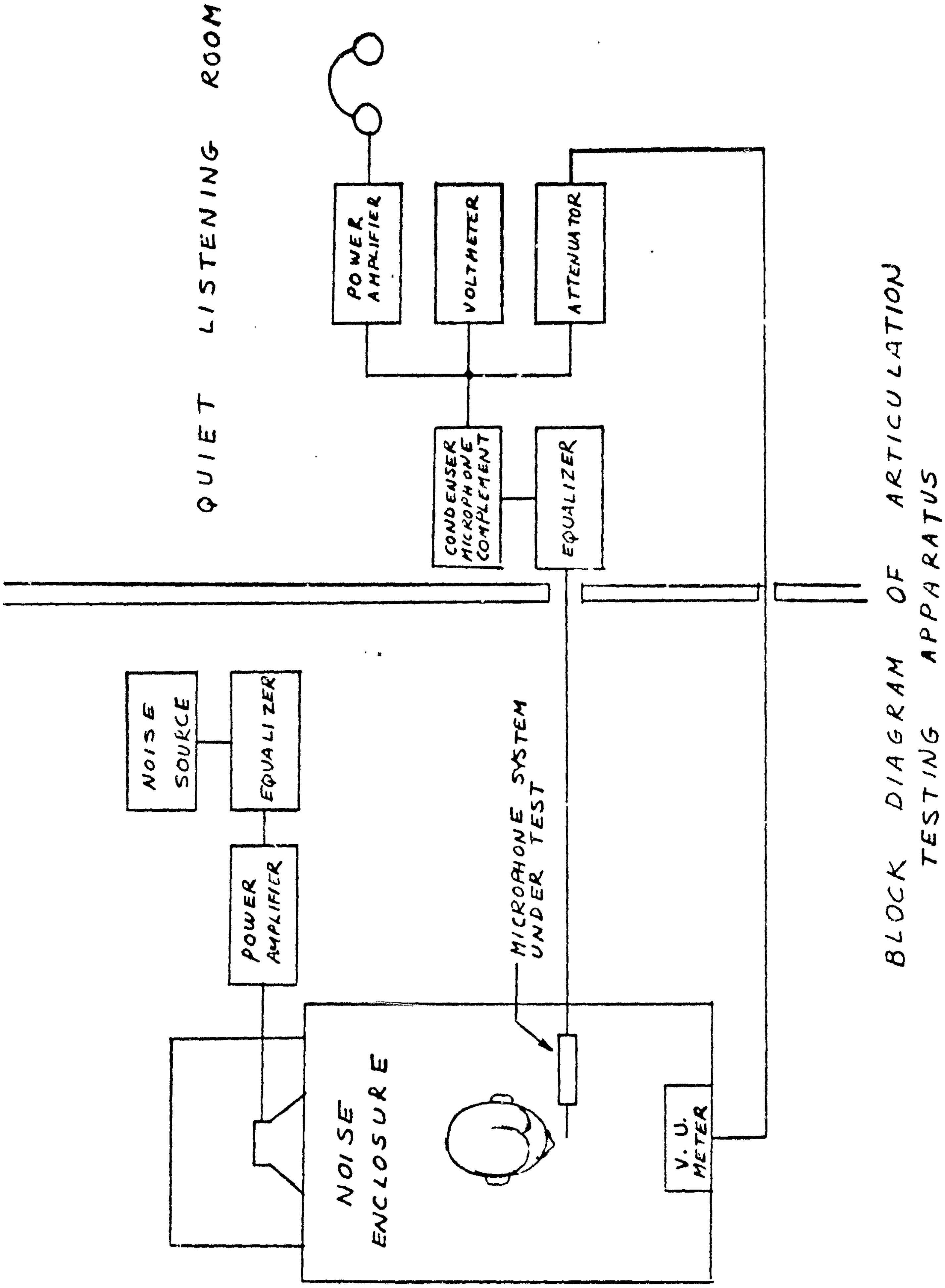


Figure A4-101

ESTIMATED MAXIMUM JET NOISE FIELD FOR 50% CONSONANT INTELLIGIBILITY.
RAISED SPEAKING EFFORT.

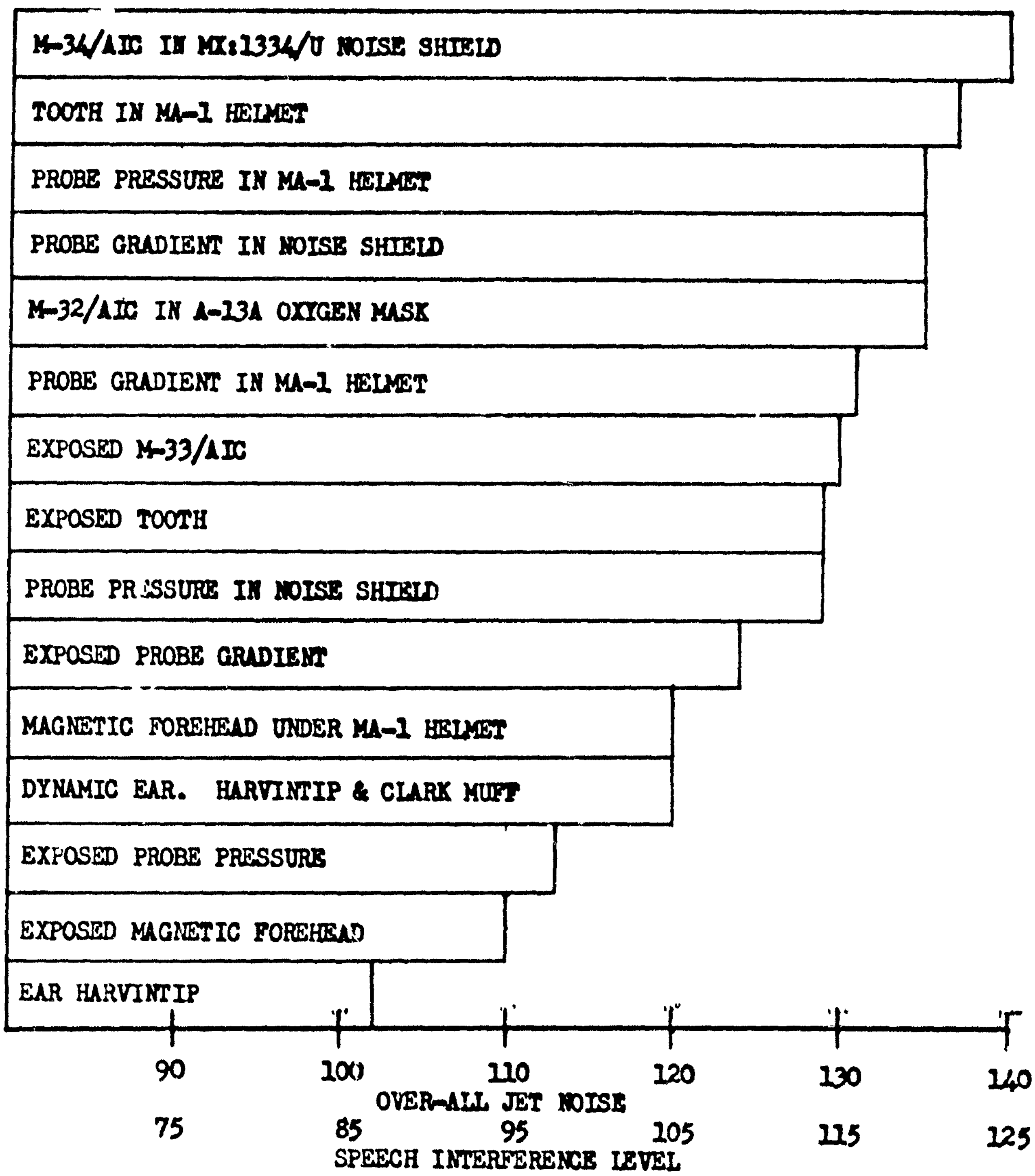


FIGURE A4-101a

BLOCK DIAGRAM OF
SPEECH ANALYSIS APPARATUS

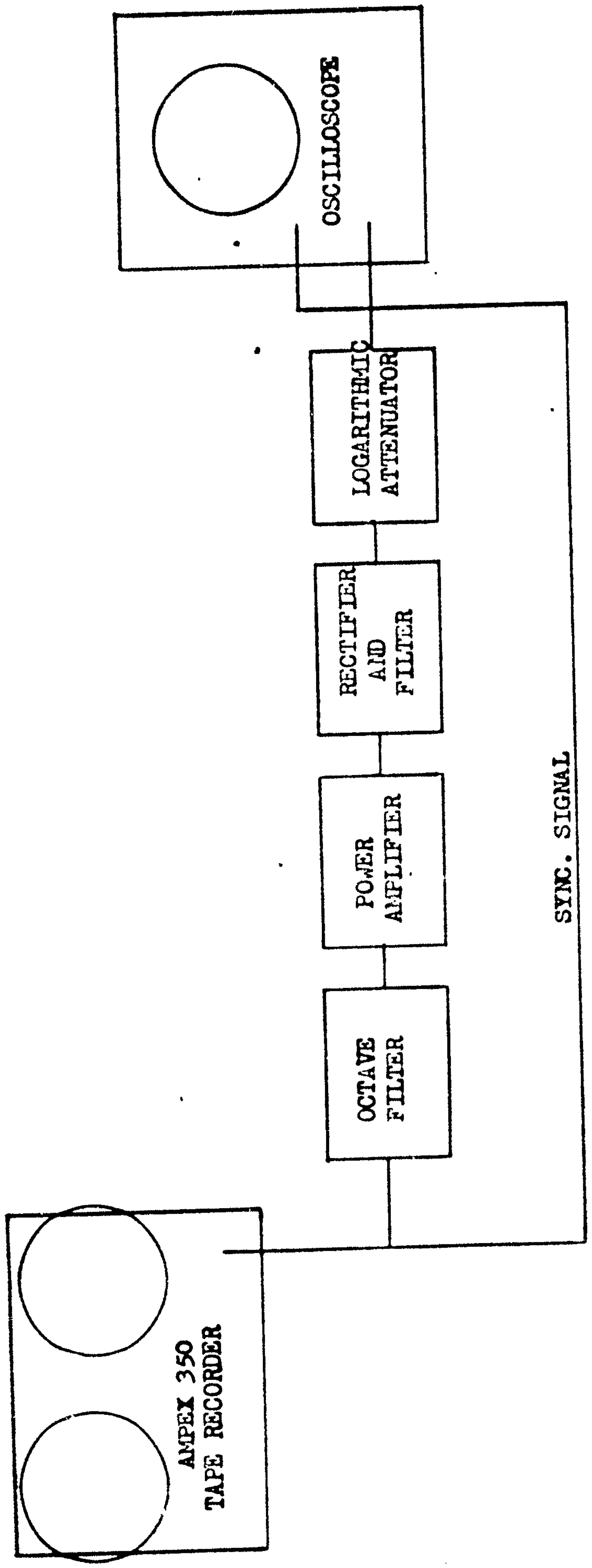
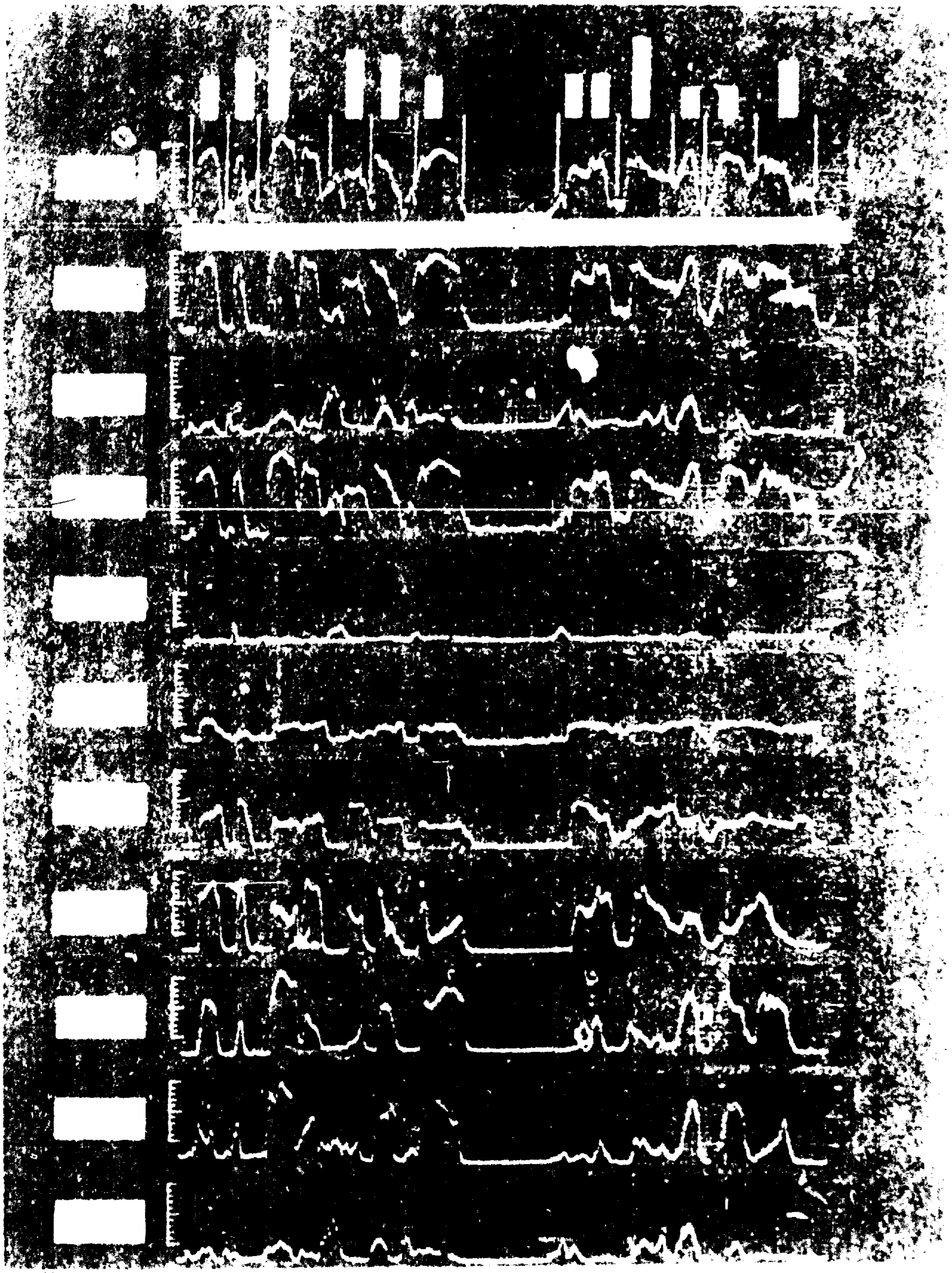
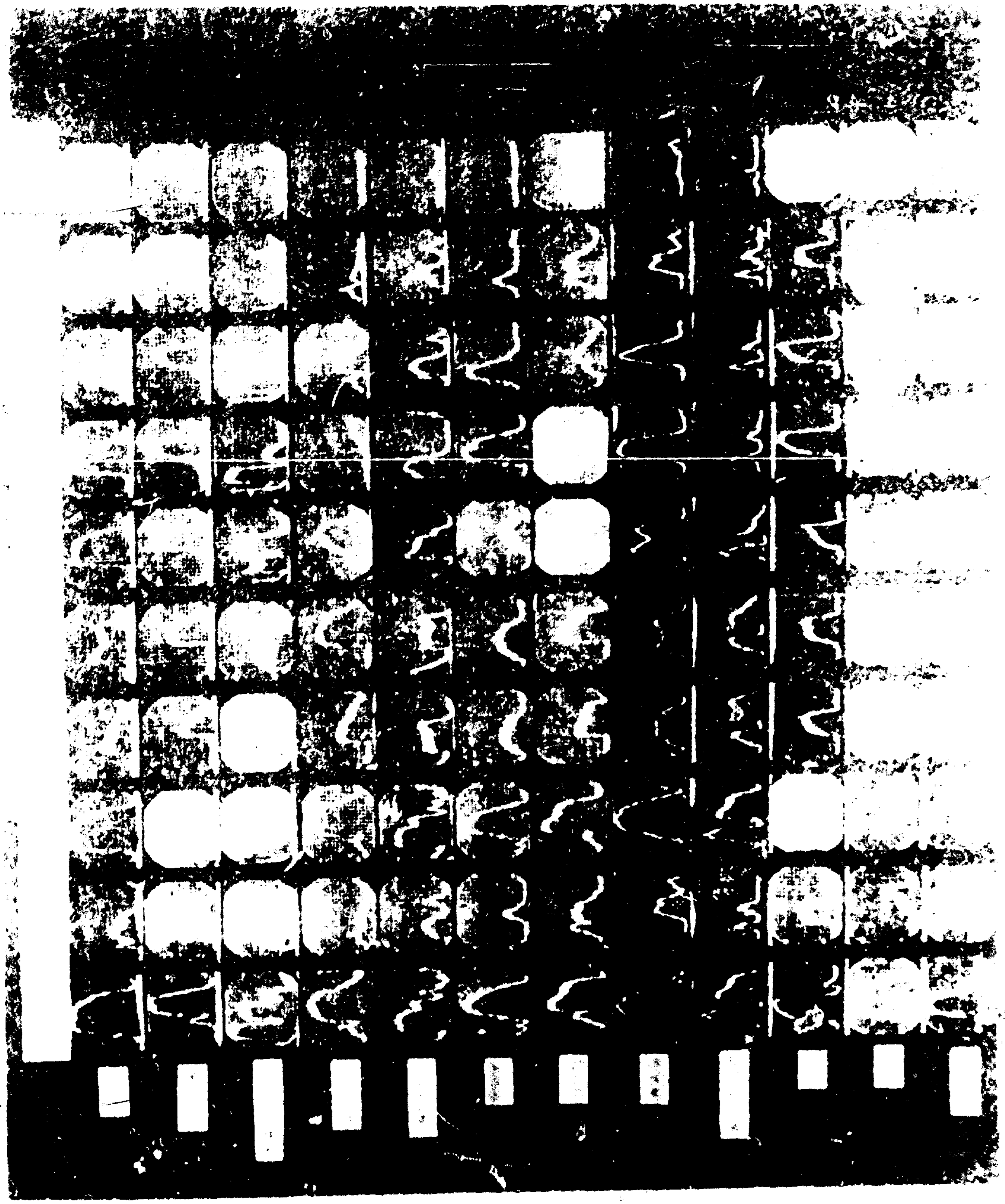
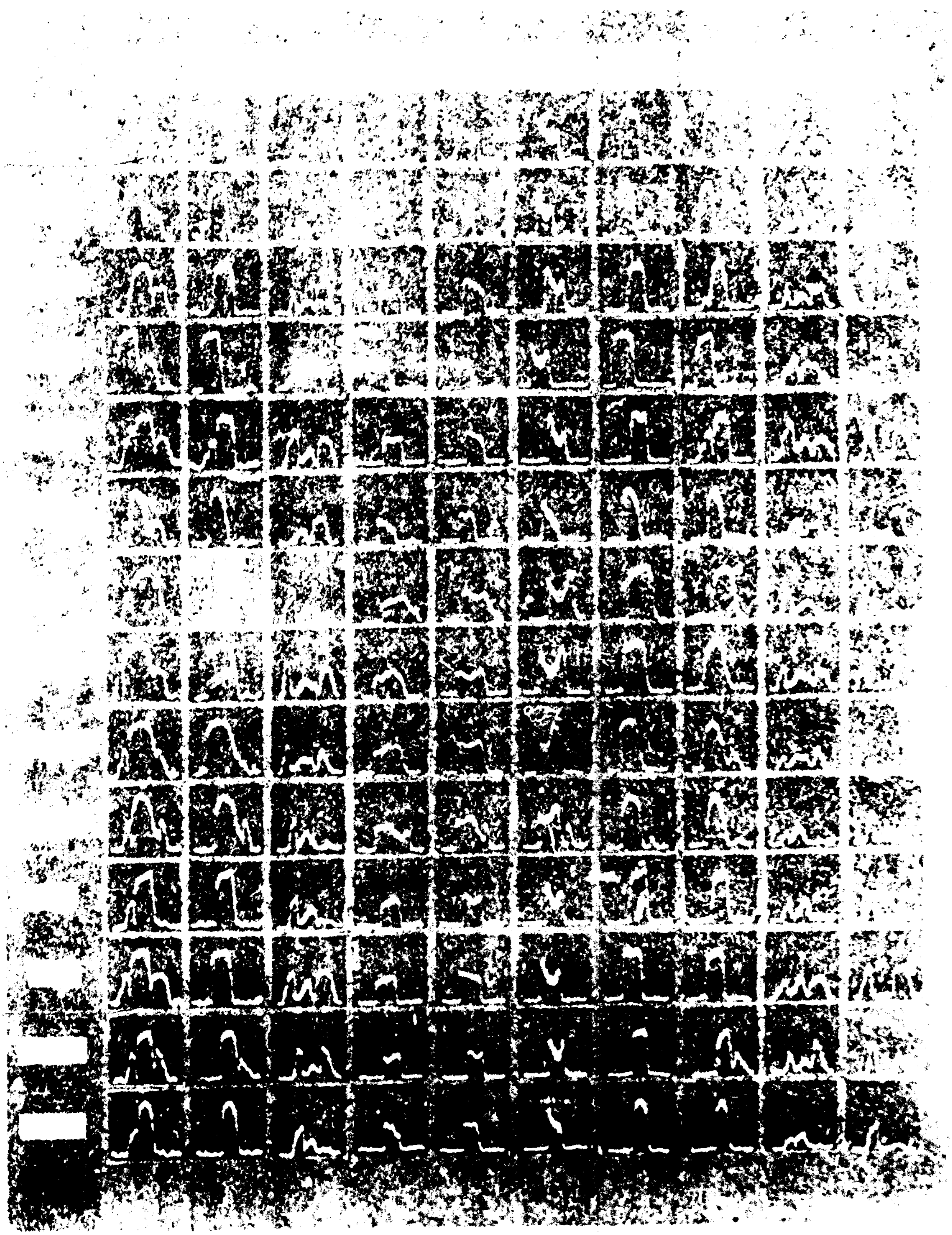


Figure A4-102



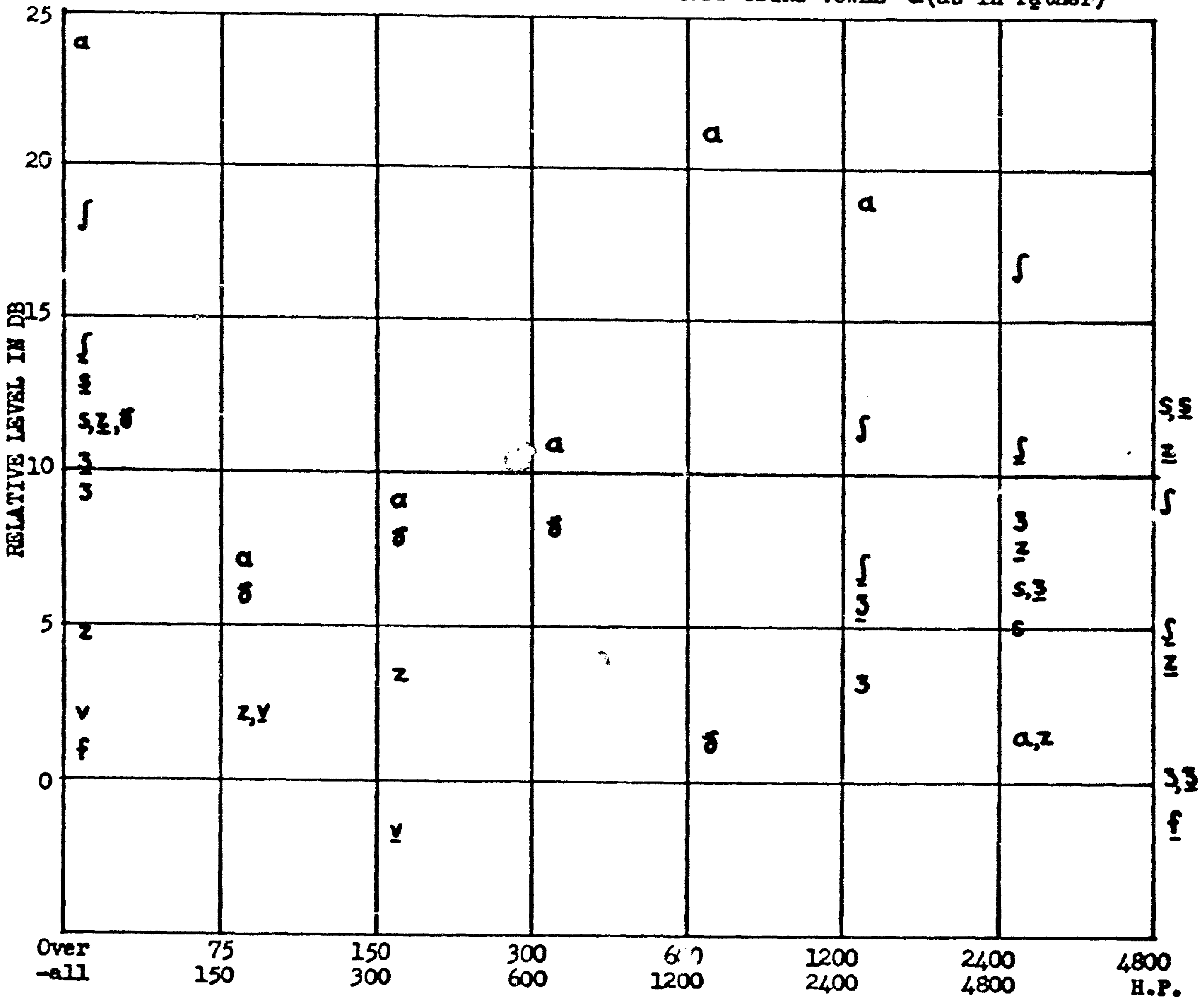
Best Available Copy





Best Available Copy

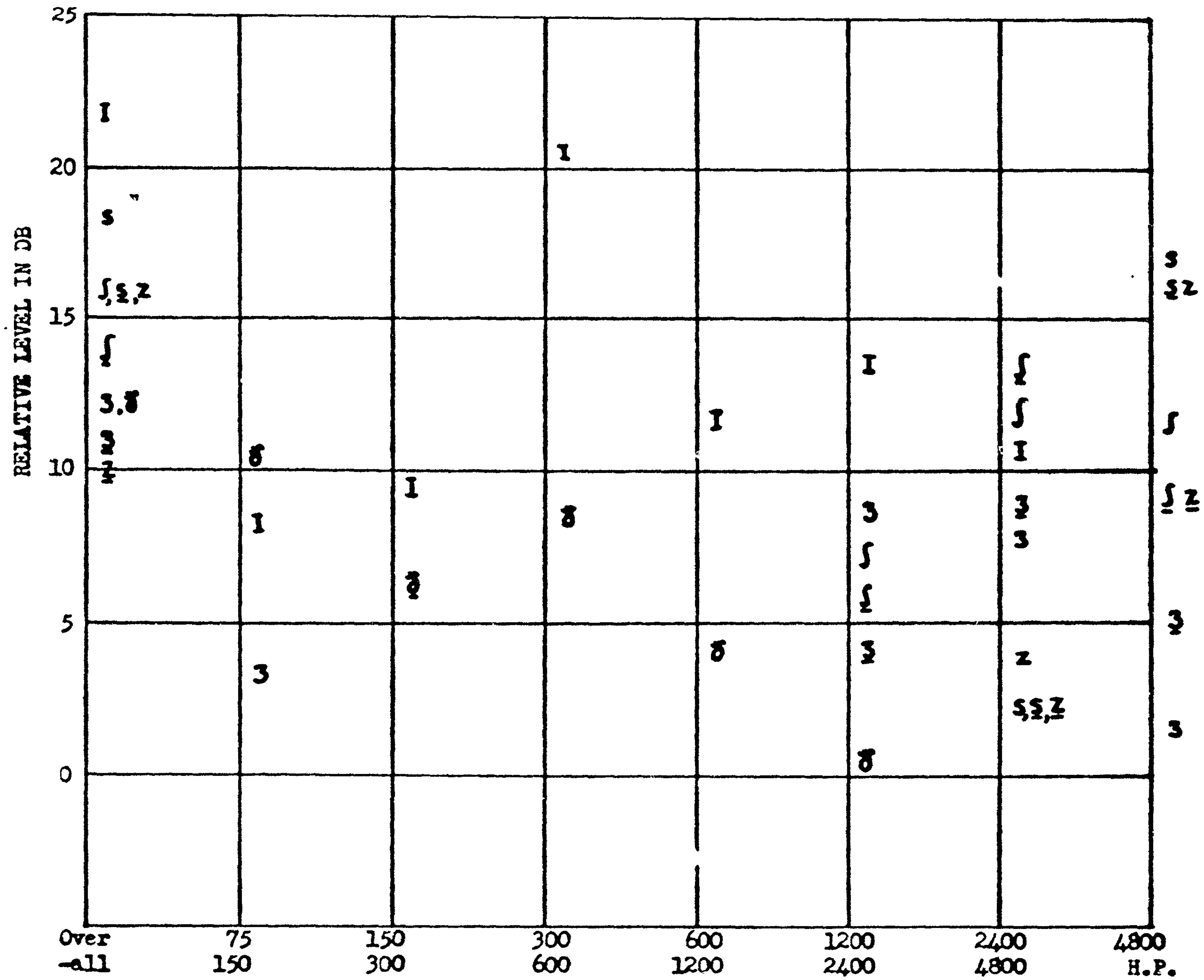
SPECTRA OF CVC WORDS USING VOWEL ɑ (as in father)



Note: International Phonetic Alphabet used. Underlining indicates the final consonant in CVC word.

Figure A4-106

SPECTRA OF CVC WORDS USING ɪ (as in s_i_t)



Note: International phonetic alphabet used. Underlining indicates the final consonant in CVC word.

Figure A4-107

Table A4 - Ia

RESULTS OF PHYSICAL MEASUREMENTS

(Averaged for Subjects TW and MG)

Apendix section	Microphone system	Improvement in S/N ratio Relative to Reference System 300/600	600/1200	1200/2400	2400/4800	4800/up	Figure of Merit R
4.2.3	Probe Pressure (reference system)	0	0	0	0	0	0
4.2.4	Probe Pressure in noise shield	30	28	24	22	20	24
4.2.5	Probe Pressure in MA-1 helmet	14	19	16	22	20	19
4.3.3	Probe Gradient	18	17	14	8	6	12
4.3.4	Probe Gradient in Noise shield	34	36	31	25	20	29
4.3.5	Probe Gradient in MA-1 Helmet	31	33	26	26	24	27
4.4.2	WE 640AA Ear (Harvin tip)	4	-2	-5	-11	-	-5
4.4.8	Dynamic Ear (Harvin tip)	1	-7	-10	-16	-	-10
4.4.8	Dynamic Ear (Harvintip + Clark Muff)	20	6	8	4	-	8
4.5	640AA Forehead	3	-2	-3	8	-	2
4.5.7	640AA Forehead (diaphram)	1	-7	3	-7	-	-2
4.5.7	Magnetic Forehead	14	6	-2	2	-	3
4.5.7	Magnetic Forehead (liquid coupling)	-10	-6	-3	-6	-	-6
4.5.7	Magnetic Forehead under MA-1 helmet	17	15	15	25	-	18
4.6	Tooth	24	12	18	21	21	19
4.6	Tooth under MA-1 helmet	35	27	28	39	40	34
4.7	M-32/AIC in MS:22001 oxygen mask	30	24	22	25	28	25
4.7	M-33/AIC	30	24	20	15	9	19
4.7	M-34/AIC in MX:1334/U Noise shield	34	32	35	37	35	35

TABLE A4-Ia

ANALYSIS OF CONFUSION AMONG SOME ENGLISH CONSONANTS*

I. Consonant bandwidth of 200 - 6500 c.p.s. for several signal to noise ratios of random noise.

(a) S/N = - 18 db

Consonant	% Heard Correctly
ð	3.2
g	4.6
ʒ	5.0
ʃ	5.2
p	5.3
θ	6.2
v	6.6
s	8.3
d	8.4
f	8.5
b	8.9
z	9.4
k	10.0
m	10.4
t	11.8
n	13.5

(b) S/N = - 12 db

Consonant	% Heard Correctly
g	12.0
ð	12.3
θ	16.5
z	16.9
t	19.5
p	19.9
d	20.6
s	22.8
b	23.4
k	26.2
v	29.4
ʒ	33.1
f	33.2
ʃ	40.4
m	51.4
n	55.8

(c) S/N = - 6 db

Consonant	% Heard Correctly
θ	22.2
ð	24.1
g	26.3
t	34.4
p	34.5
d	37.0
k	37.8
z	39.2
s	43.2
v	54.2
b	54.8
ʒ	55.6
f	60.8
n	69.1
m	75.0
ʃ	78.6

(d) S/N = 0 db

Consonant	% Heard Correctly
θ	49.1
k	50.0
p	50.7
ð	51.9
z	54.0
g	59.1
d	72.6
v	75.0
f	75.3
t	76.5
s	78.8
b	85.2
ʒ	91.8
n	95.2
m	96.6
ʃ	96.7

(e) S/N = + 6 db

Consonant	% Heard Correctly
θ	62.2
ð	67.2
p	68.6
g	70.3
v	75.0
f	75.0
k	79.1
b	82.3
z	83.8
d	88.7
t	92.4
s	95.0
ʒ	95.9
m	98.6
n	98.9
ʃ	99.9

(f) S/N = + 12 db

Consonant	% Heard Correctly
p	62.5
θ	69.0
v	82.1
g	84.6
b	86.4
ð	86.4
f	87.9
d	90.2
k	91.2
z	94.5
t	98.5
m	99.3
s	100.0
ʃ	100.0
ʒ	100.0
n	100.0

* From Miller and Nicely, J. Acoust. Soc. Am. 27, 338 - 1955

Table A4-I

II. Constant signal to noise ratio of 12 db

High pass cut off fixed at 200 c.p.s. with low pass cut off of 300, 400, 600, 1200. 2500. 5000.

(a) Frequency response of 200 - 300 c.p.s.

Consonant	% Heard Correctly
g	12.1
ʒ	14.5
θ	16.5
f	16.7
ʃ	16.8
ð	19.1
p	19.3
t	23.5
s	25.8
z	27.1
v	28.7
d	30.0
k	33.7
b	38.9
n	40.2
m	64.9

(b) Frequency response of 200 - 400 c.p.s.

Consonant	% Heard Correctly
ð	24.6
p	25.7
g	25.9
t	27.3
f	29.2
ʒ	31.0
θ	34.2
z	37.3
ʃ	40.5
s	43.4
k	43.5
v	47.9
n	50.0
b	55.9
d	61.3
m	75.0

(c) Frequency response of 200 - 600 c.p.s.

Consonant	% Heard Correctly
t	29.7
ð	34.1
ʃ	34.6
s	35.8
θ	37.9
g	40.4
ʒ	40.8
z	46.8
p	47.1
n	50.0
f	50.8
k	54.0
v	59.8
d	64.0
b	81.0
m	84.0

(d) Frequency response of 200 - 1200 c.p.s.

Consonant	% Heard Correctly
t	33.0
θ	33.4
ð	34.8
s	35.2
ʃ	37.5
z	46.7
g	46.9
ʒ	53.4
k	57.4
f	59.0
v	59.2
p	66.5
d	75.8
b	88.5
m	95.2
n	98.0

(e) Frequency response of 200 - 2500 c.p.s.

Consonant	% Heard Correctly
θ	37.5
t	43.0
ð	50.0
s	52.3
z	52.6
v	66.5
g	72.9
p	77.9
f	78.8
d	84.6
b	85.9
k	86.7
ʃ	89.0
ʒ	89.5
m	93.0
n	99.2

(f) Frequency response of 200 - 5000 c.p.s.

Consonant	% Heard Correctly
θ	56.2
ð	60.7
z	69.7
v	70.0
b	78.4
s	78.4
f	79.5
g	81.6
k	87.4
p	93.4
ʒ	94.3
m	95.0
d	95.2
t	96.6
ʃ	99.5
n	100.0

Table A4-I (continued)

III. Constant signal to noise ratio of 12 db

Low pass cut off fixed at 1000 with high pass cut off of 1000, 2000, 2500, 3000, 4500 c.p.s.

(a) Frequency response of 1000 - 5000 c.p.s.

Consonant	% Heard Correctly
ð	46.0
b	47.8
v	50.3
θ	53.9
z	62.6
f	64.3
g	67.4
d	71.6
p	73.4
s	7[illegible].0
m	87.5
ʒ	89.3
n	90.8
k	91.4
t	98.4
ʃ	98.5

(b) Frequency response of 2000 - 5000 c.p.s.

Consonant	% Heard Correctly
v	18.7
θ	22.8
ð	24.6
b	25.4
f	26.9
p	38.8
z	42.7
m	50.5
k	54.4
s	57.0
d	70.5
g	71.5
n	75.0
ʒ	81.6
ʃ	86.8
t	87.2

(c) Frequency response of 2500 - 5000 c.p.s

Consonant	% Heard Correctly
f	10.7
v	13.8
ð	15.5
b	18.2
θ	19.4
p	24.0
m	27.0
ʃ	27.2
k	31.6
z	35.2
g	35.4
n	46.5
s	54.8
d	57.2
ʒ	68.5
t	82.0

(d) Frequency response of 3000 - 5000 c.p.s.

Consonant	% Heard Correctly
b	11.5
θ	13.0
p	14.1
z	16.3
g	16.8
k	21.6
m	22.3
n	23.0
d	24.3
ʒ	50.8
s	52.2
t	66.6
v	68.7
f	69.5
ð	74.0
ʃ	75.0

(e) Frequency response of 4500 - 5000 c.p.s.

Consonant	% Heard Correctly
n	6.9
m	7.1
θ	7.3
p	10.3
ð	10.8
b	11.0
v	12.7
z	20.6
ʒ	31.7
s	48.6
t	59.8
ʃ	69.9
d	73.8
f	81.9
g	87.5
k	95.0

Table A4-I (continued)

RANK ORDER OF SIXTEEN CONSONANTS IN ORDER OF DIFFICULTY

(In the chart below, "1" means that this consonant was most frequently confused, while "16" means that it was the least frequently confused.) From Miller and Nicely, J. Acoust. Soc. Am. 27, 338 - 1955

Consonant	SIGNAL TO NOISE RATIO: S/N -18 db	S/N -12 db	S/N -6 db	S/N 0 db	S/N 6 db	S/N 12 db	BANDWIDTH: 200-300 cps	200-400 cps	200-600 cps	200-1200 cps	200-2500 cps	200-5000 cps	1000-5000 cps	2000-5000 cps	2500-5000 cps	3000-5000 cps	4500-5000 cps
ð	1	2	2	4	2	6	6	1	2	3	3	2	1	3	3	15	5
g	2	1	3	6	4	4	1	3	6	7	7	8	7	12	11	5	15
ʒ	3	12	12	13	13	15	2	6	7	8	14	11	12	14	15	10	9
ʃ	4	14	16	16	16	14	5	9	3	5	13	15	16	15	8	16	12
p	5	6	5	3	3	1	7	2	9	12	8	10	9	6	[illegible]	3	4
θ	6	3	1	1	1	2	3	7	5	2	1	1	4	2	5	2	3
v	7	11	10	8	5	3	11	12	13	11	6	4	3	1	2	13	7
s	8	8	9	11	12	13	9	10	4	4	4	6	10	10	13	11	10
d	9	7	6	7	10	8	12	15	14	13	10	13	8	11	14	9	13
f	10	13	13	9	6	7	4	5	11	10	9	7	6	5	1	14	14
b	11	9	11	12	8	5	14	14	15	14	11	5	2	4	4	1	6
z	12	4	8	5	9	10	10	8	8	6	5	3	5	7	9	4	3
k	13	10	7	2	7	9	13	11	12	9	12	9	14	9	10	6	16
m	14	15	15	15	14	12	16	16	16	15	15	12	11	8	7	7	2
t	15	5	4	10	11	11	8	4	1	1	2	14	15	16	16	12	11
n	16	16	14	14	15	16	15	13	10	16	16	16	13	13	12	8	1

Table A4-II

RANK ORDER OF SIXTEEN CONSONANTS IN ORDER OF DIFFICULTY ON THE BASIS OF SIGNAL TO NOISE RATIO. FROM MILLER AND NICELY, J. ACOUST. SOC. AM. 27, 338 - 1955.

order of difficulty	consonant
1	θ
2	ð
3	g
4	p
5	v
6	d
7	z
8	k
9	t
10	b
11	f
12	s
13	ʒ
14	m
15	ʃ
16	n

Table A4-III

CONSONANTS USED IN WESTERN ELECTRO-ACOUSTIC LABORATORY
ARTICULATION TESTS

Consonant	Occurance in Speech[1]	Confusion in Noise: First Susceptible to Noise[2]	Confusion in Noise: Overall Difficulty[3]	Classification
p	11	1	4	Voiceless Stop
t	1	11	9	Voiceless Stop
k	5	9	8	Voiceless Stop
f	9	7	11	Voiceless Fricative
θ	15	2	1	Voiceless Fricative
s	4	13	12	Voiceless Fricative
b	16	5	10	Voiced Stop
d	3	8	6	Voiced Stop
g	12	4	3	Voiced Stop
v	10	3	5	Voiced Fricative
z	8	10	7	Voiced Fricative
ʒ	7	6	2	Voiced Fricative
m	6	12	14	Nasal
n	2	16	16	Nasal

1. H. Fletcher, Speech and Hearing in Communication (D. Van Nostrand Company, Inc. 1953 p. 96)

2. Miller and Nicely, J. Acoust. Soc. Am. 27, 342, 1955 (Table VI)

3. From an analysis of Reference 2 by Western Electro-Acoustic Laboratory, Inc.

Table A4-IV

TABLE A4-V

Summary of Articulation Testing

System	# Speak. 1	# List. 2	No. Max. 3	Avg. Score 4	S/N	DB Below Max. 5	DB Below Max. for 50% Score 6
WEAL Press. Open	21	40	7	53	-10	20	20
WEAL Grad. Open	29	41	9	53	-10	31	31
WEAL Press. N.S.	18	24	6	68	0	32	36
WEAL Grad. N.S.	14	26	7	76	0	36	42
WEAL Press. MA-1	6	12	2	52	0	36	36
WEAL Grad. MA-1	6	12	2	61	0	36	38
MAG. for Open	9	18	3	45	-8	18	17
MAG. for MA-1	9	18	3	42	-6	28	27
Ear - Open	9	18	3	68	-7	5	9
Ear - Clark	9	18	3	47	-8	27	27
Tooth - Open	15	30	5	63	--6	33	36
Tooth - MA-1	9	18	3	82	2	34	44
M-32	16	30	6	82	0	32	42
M-33	18	25	6	68	-10	33	37
M-34	24	28	6	83	0	37	47

1. Total number of speaking tests.
2. Total number of listening tests.
3. Number of times maximum effort was established.
4. Consonant articulation score $= \frac{\text{Total number correct}}{\text{Total Number}} \times 100\%$
5. Decibels below maximum effort.
6. Data of (5) extrapolated to 50% using Figure A4-100.

TABLE A4-V

TABLE A4-VI

Rank Order of Microphone Systems on the Basis of Articulation Testing.

(1) System	(2) DB Below Max. Effort for 50% Score	(3) DB Below Max. Effort for 50% Score	(4) Predicted Max. S/N Environment for 50% CVC Score at Raised Effort *
M-34 in MX:1334/J Shield	47	20	140
Tooth in MA-1 Helmet	44	17	137
M-33 in A-13A Mask	42	15	135
WEAL Grad. in Noise Sh.	42	15	135
WEAL Grad. in MA-1 Helmet	38	11	131
M-33 (exposed)	37	10	130
WEAL Press. in MA-1	36	9	129
WEAL Press. in Noise Sh.	36	9	129
Tooth (exposed)	36	9	129
WEAL Grad. (exposed)	31	4	124
Mag. Forehead-MA-1	27	0	120
Ear - Clark Muff	27	0	120
WEAL Press. (exposed)	20	-7	113
MAG. Forehead (exposed)	17	-10	110
Ear (exposed)	9	-18	102

*Figures given are overall jet noise in db re .0002 dyne/cm^2 of assumed spectrum shape of Figure 12. The SIL is 15 db below this figure.

TABLE A4-VI

TABLE A-4 VII

Difference Between Maximum Effort and

Other Subjective Efforts, in decibels.

Speech Effort	LTA Overall SPL at one foot	Difference Relative to Maximum Effort
Maximum	102	0
Very Loud	84	18
Raised	75	27
Normal Conversation	66	36
Lowered	60	42
Very Soft	55	47
Whisper	44	58

SUMMARY OF THE EVALUATION OF MICROPHONE SYSTEMS ON THE BASIS OF SPEECH SOUND ALTERATION, QUALITY, AND SPEAKER RECOGNIZABILITY.

SYSTEM	SPEECH SOUND ALTERATION				QUALITY	RECOGNIZABILITY
	VOWELS	FRICATIVES	PLOSIVES	NASALS		
M-32/AIC in HS:22001 Ox. Mask	nat	accen	nat	nat	oral metallic	poor
M-33/AIC in open	nat	nat	nat	dim	denasal metallic	fair
M-34/AIC in MX:1334/U noise shield		nat	accen	nat	nasal metallic	poor
W.E. 640AA Pressure Probe Mic.	nat	nat	nat	nat	nat	good
W.E. 640AA Pressure Probe Mic. in WEAL fiberglas noise shield	nat	nat	nat	nat	nat	good
W.E. 640AA Probe Gradient Mic.	nat	accen	accen	dim	denasal	good
W.E. 640AA Probe Gradient Mic. in WEAL fiberglas noise shield	nat	nat	nat	nat	metallic	good
W.E. 640AA Probe Gradient Mic. in MA-1 Helmet						
W.E. 640AA Ear Mic. (Harventip insert tip)	nat	sl.dim	nat	nat	bass	good
Dynamic Ear Mic. (Harventip insert tip)	nat	sl.dim	nat	nat	bass	good
W.E. 640AA forehead microphone	nat	dim	nat	nat	bass	fair
Magnetic Forehead Mic.	nat	dim	nat	nat	bass	fair
Tooth Mic.	nat	accen	altered	nat	metallic	fair

Table A4-VIII

VOWELS AND DIPTHONGS

KEY WORD	PHONETIC SYMBOLS AUTHORITY IPA,	HARV,	BELL,	WEBS,	CLASSIFICATION,	FREQUENCY OF OCCURANCE %.	RANK,	REL. PHONETIC POWER DB.	RANK.	CONFUSION IN NOISE.
see	i	ē	ē	ē	front vowel	2.1	5	23.4	11	8
city	i			ė	"	-	-	-	-	-
sit	I	i	i		"	6.3	1	24.1	10	6
April	I			ĭ	"	-	-	-	-	-
set	ɛ	e	e	ĕ	"	2.7	3	25.4	8	7
sat	æ	á	á	ă	"	2.1	6	26.9	4	9
half	a			ȧ	"	-	-	-	-	-
father	a	a	a	ä	central vowel	2.1	7	27.8	2	12
swap	ɒ			ŏ	"	-	-	-	-	-
cut	ʌ	u	o	ŭ	back	6.0	2	27.1	3	5
all	ɔ	o	ó	ô	"	1.3	11	28.3		11
could	U	u	U	ŏŏ	"	1.0	12	26.6	6	3
boot	u	ū	ū	ōō	"	2.0	8	24.9	9	1
bird[illegible]	ɜ			û	central vowel	-	-	-	-	-
bird[illegible]	ɝ			û	"	-	-	-	-	-
bottle	l̩	l	l	l	semi vowel	-	-	-	-	-
sofa	ə			ȧ	schwa vowel	-	-	-	-	-
time	aI	ī	ī	ī	dipthong	2.4	4	-	-	-
sound	aU	ow	ou	ou	"	0.6	13	-	-	-
boil	I	oi	oi	ei	"	0.1	14	-	-	-
few	ju	ew	ew	ū	"	0.1	15	-	-	-
say	eI	a	a	ā	"	1.9	9	25.7	7	10
no	oU	ō	ō	ō	"	1.5	10	26.7	5	2

Table A4-IX

VOWELS

1. International Phonetic Alphabet.

2. Symbols used in J.P. Egan, "Articulation Testing Methods II," OSRD Report 3802, Psycho-Acoustic Laboratory, Harvard University, Cambridge (1944).

3. Bell Telephone Laboratory Symbols. See H. Fletcher, Speech and Hearing, D. Van Nostrand, New York, 1953, p.3.

4. Symbols used in Webster's New-International Dictionary.

5. Kanter and West, Phonetics. Harpers, 1941, p. 178-179 and ref. 3 p.2.

6. Ref. 3, p.96, Table 15. Conversation Speech Only. Results are given in % occurance considering all speech sounds.

7. Based on results of 6. The vowels have been ranked on basis of occurance. 1 indicates most frequently occuring vowel, etc.

8. Ref. 3, p.86, Table 7A. The results have been converted to db. 0 db corresponds to the least intense sound (as in thin).

9. Based on results of 8. 1 indicates the most intense vowel, etc.

10. J.M. Pickett, J. Acoust. Soc. Am. 29, 613 (1957). Based on date from Table II, "flat noise" only. 1 indicates vowel u (as in boot) was the most difficult to perceive in "flat" noise with a S/N of -11 db, -15 db, etc.

Table A4-IX (continued)

CONSONANTS

KEY WORD	PHONETIC SYMBOLS AUTHORITY IPA[1]	HARV[2]	BELL[3]	WEBS[4]	CLASSIFICATION[5]	FREQUENCY OF OCCURANCE %[6]	RANK[7]	REL. PHONETIC POWER DB[8]	RANK[9]	CONFUSION IN NOISE[10]
those	ð	th'	th	th	voiced fric.	2.5	10	10.4	16	2
vat	v	v	v	v	"	1.8	15	10.8	15	5
zero	z	z	z	z	"	2.2	11	12.0	10	7
azure	ʒ	zh	zh	zh	"	.01	24	13.0	9	13
fan	f	f	f	f	voiceless fric.	2.0	14	7.0	20	11
say	s	s	s	s	"	4.0	6	12.0	11	12
ship	ʃ	sh	sh	sh	"	0.7	19	19.0	3	15
thin	θ	th	th	th	"	0.7	20	0	21	1
boat	b	b	b	b	voiced stop (plosive)	0.6	21	8.5	17	10
do	d	d	d	d	"	4.6	5	8.5	18	6
go	g	g	g	g	"	1.5	17	12.0	13	3
judge	dʒ	j	j	j	"	0.3	22	13.6	8	-
chose	tʃ	ch	ch	ch	voiceless stop (plosive)	0.3	23	16.2	6	-
king	k	k	k	k	"	3.6	8	11.4	14	8
to	t	t	t	t	"	9.8	1	11.7	12	9
pin	p	p	p	p	"	1.7	16	7.8	19	4
man	m	m	m	m	semi vowel Nasal	3.6	12	17.2	5	14
no	n	n	n	n	"	8.1	2	15.6	7	16
sing	ŋ	ng	ng	ng	"	1.1	18	18.6	4	-
lady	l	l	l	l	"(reced glide)	4.6	4	20	2	-
run	r	r	r	r	" " "	6.1	3	23.2	1	-
win	w	w	w	w	transitional (app. glide)	3.7	7	-	-	-
what		hw			"	-	-	-	-	-
you	j	y	y	y	"	2.1	13	-	-	-
hot	h	h	h	h	"(glottal fric.)	2.2	12	-	-	-

Table A4-X

CONSONANTS

1. International Phonetic Alphabet.

2. Symbols used in J.P. Egan, "Articulation Methods II," OSRD Report 3802, Psycho-Acoustic Laboratory, Harvard University, Cambridge (1944).

3. Bell Telephone Laboratory Symbols. See H. Fletcher, Speach and Hearing, D. Van Nostrand, New York, 1953. p.3.

4. Symbols used in Webster's New International Dictionary.

5. Kanter and West. Phonetics. Harper's, 1941, p. 178-179 and ref. 3 p.2.

6. Ref. 3, p.96, Table 15. Conversational Speech Only. Results are given in % occurance considering all speech sounds.

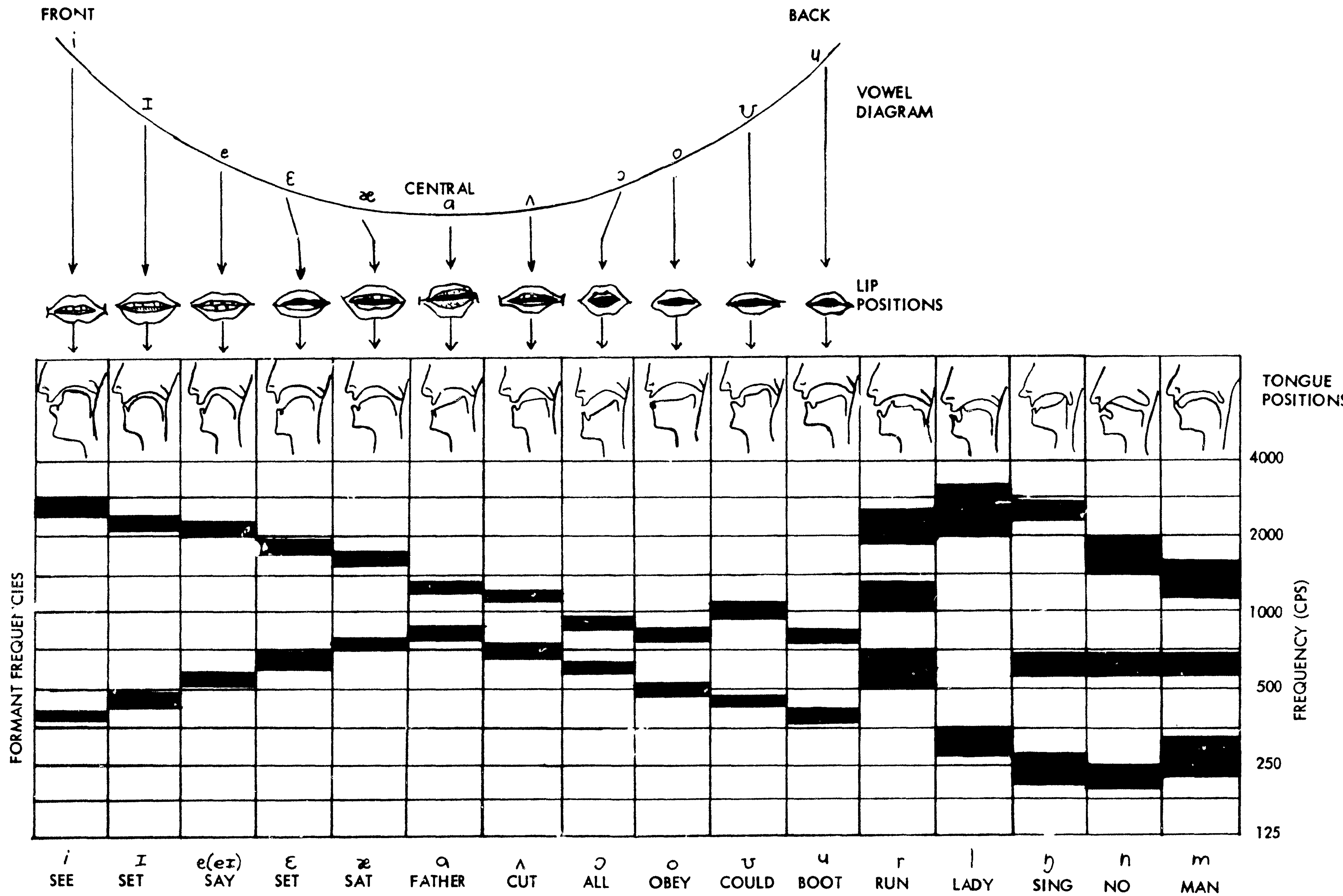
7. Based on results of preceding column (ref. 6). The consonants have been ranked on basis of occurance. One indicates the most frequently occuring consonant, etc.

8. Ref. 3, p.86, Table 7A. The results were converted to db. 0 db corresponds to the least intense sound (as in thin).

9. Based on results of preceding column (ref. 8). One indicates the most intense consonant, etc.

10. Miller and Nicely. J. Acoust. Soc. Am. 27,342,1955 (Table VI). These results were analyzed (see appendix section) by WEAL to obtain a difficulty rating for the consonants. One indicates the most difficult consonant to perceive, etc.

Table A4-X (continued)


FRONT
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VOWEL DIAGRAM
LIP POSITIONS
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FORMANT FREQUEI 'CIES
4000
2000
1000
500
250
125
FREQUENCY (CPS)
i SEE
I SET
e(eɪ) SAY
ɛ SET
æ SAT
a FATHER
ʌ CUT
ɔ ALL
o OBEY
ʊ COULD
u BOOT
r RUN
l LADY
ŋ SING
n NO
m MAN

TABLE A4 - X1

APPENDIX 5.0

STUDY OF SPEECH RECEPTION TYPES

BY

RADIO CORPORATION OF AMERICA

APPENDIX 5

STUDY OF SPEECH RECEPTION TYPES

OUTLINE OF FIGURES IN APPENDIX 5

APPENDIX 5

5.1 AN/AIC-10 TYPES AND PRIOR HEADSETS

For the purposes of this study it will only be necessary to discuss a few typical models in this category. Information regarding some of the devices was extracted from existing reports and memoranda. For those items, which were evaluated by this laboratory, standard testing procedures were employed, as discussed in Appendix 5.6.

1. Prior Headsets

A. HS-33

Information regarding this headset was obtained from References 1, 2, and 3. The nomenclature of the earphone used in this headset is ANB-H-1. Its physical dimensions are as follows:

1. Thickness	1 inch
2. Diameter	2 inches
3. Weight	73 grams

It was made in two basic designs, one a magnetic diaphragm type and the other a moving armature type both having an inductive impedance at 1000 cps of approximately 300 ohms. Response characteristics as measured with a 6cc coupler for both ground and altitude are shown in Figure A5-1. Sound attenuation using an MC-162A earcushion is shown in Figure A5-2. The MC-162A earcushion is relatively small and made of hard rubber. It couples to the ear by resting directly on the pinna. Probe tube measurements of the pressure response of the combination, ANB-H-1 and MC-162A on the ear are shown in Figure A5-3, taken from Reference 3. The earcushion was later changed to the M-301, a circumaural type. Unfortunately, information regarding this earcushion could not be located in time for inclusion in this report. However, information regarding the Harvard Design 5-B and 6 earcushions with the ANB-H-1 was available. It is believed that these cushions can be used as being representative of the M-301 earcushion. Probe tube measurements of the pressure response and attenuation curves are shown in Figures A5-4 and A5-5 respectively.

Articulation calculations for each of the above combinations were performed by the methods given in Appendix G. The noise spectrum used for these calculations and all the preceding ones was the W.E.A.L. Spectrum as shown in Figure A5-6. Articulation calculations are shown in Figures A5-7 and A5-8.

B. Permoflux, PDR-3

References 2 and 3 were used to obtain all the necessary information on this headset required for this report. The earphone used in this headset, the ANB-H-1A was mechanically interchangeable with the ANB-H-1. Its physical characteristics are shown in the following table:

1.	Thickness	1 inch
2.	Diameter	2 inches
3.	Weight	112 grams

It was of the moving coil type and had an essentially resistive impedance of 300 ohms. The 6cc coupler response for the earphone as a function of altitude is shown in Figure A5-9. Real-ear response and sound attenuation curves are shown in Figures A5-10 and A5-11 using the MC-162A earcushion. The real-ear response and sound attenuation of the ANB-H-1A in the Harvard cushions are shown in Figures A5-12 and A5-13. Calculation of word articulation is shown in Figures A5-14 and A5-15.

2. H-70/AIC Headset (Original - Reference 8)

The primary acoustic components of the headset are the H-79/AIC receiver and a three ring fiberglass earcushion. The receiver is a moving coil low impedance earphone. It differs from previous types used in that the diaphragm motion is predominantly controlled by acoustic stiffness over a wide portion of the operating range. A small cavity, located behind the diaphragm, provides an acoustic stiffness impedance which has considerably greater effect at both ground level and elevated altitude than the mechanical stiffness of the mechanical system. The output level is maintained nearly constant, independent of air density variations from ground level to 40,000 feet altitude. The physical characteristics of the H-79/AIC telephone receiver are as follows:

Diameter	2 inches
Height	1/2 inch
Weight	80 grams

It has a resistive impedance of approximately 15 ohms. The earphones are connected in parallel in the headset resulting in a headset impedance of 7.5 ohms. The 6cc coupler response as a function of altitude is shown in Figure A5-16.

The earcushion consists of three one-inch thick rings of superfine fiberglass, compressed between a dipped neoprene latex skin, .014 inch thick, and a molded nylon backing shell. The front face of the earcushion is covered by a nylon cloth cover.

Real-ear response for the H-70/AIC headset is shown in Figure A5-17. The method of evaluation consisted of 1000 cps loudness comparison at a 100 phon loudness level (Reference 9). The sound attenuation is shown in Figure A5-18. Calculation of articulation index is shown in Figure A5-19. This value differs from that calculated for the H-70/AIC headset in Appendix 57. The reason for this difference is that in Appendix 57 the RCA noise spectrum was used and in this case the W.E.A.L. spectrum was used.

3. H-70/AIC Headset

The primary acoustic components of this headset are the H-79/AIC receiver and the MX-2088/U earcushion. Although the earphone in this headset has the same designation as that of the H-70/AIC headset, it has increased output of about 6 db. Frequency response of this earphone as a function of altitude is shown in Figure A5-20. It is mechanically interchangeable with the original model. It too is a moving coil type. The earphone has a nominal resistive impedance of 19.5 ohms; headset impedance is thus about 10 ohms. It has the same physical characteristics as the H-79/AIC in the H-70/AIC headset.

The earcushion is appreciably different. It is a circumaural foam rubber cushion, hypalon coated and conforms to the U.S.A.F. Drawing No. 56E 12858.

Real-ear response as obtained by loudness balance and real-ear attenuation at threshold are shown in Figures A5-21 and A5-22 respectively. The calculation of the articulation index is shown in Figure A5-23.

4. H-158 (Reference 10)

The H-158 is electrically interchangeable with the H-70/AIC headset, but has vastly improved acoustical and mechanical properties. The earphone, H-143, developed under Contract AF 33(600)-33172, is comparable to the H-79 receiver with regard to sensitivity and altitude performance (Figure A5-24), however, its weight is appreciably less than the H-79. The physical dimensions are as follows:

Height	less than 1/2 inch
Diameter	2 inches
Weight	35 grams

The earphone is enclosed in a rigid cycolac shell which has a volume of approximately 100 cc. A flange is provided on the shell to accommodate a vinyl pad containing either a glycerine-base liquid or isocyanate foam filler. Attenuation measurements made using both types of pads are shown in Figures A5-25 and A5-26. There is little difference between the two curves and at present the isocyanate foam pad is preferred, from the standpoint of mechanical reliability. Real-ear response measurements made by loudness balance techniques for either the liquid or foam pad

are shown in Figure A5-27. In addition, real-ear response measurements at threshold for the H-157 with either pad were determined, the results are shown in Figure A5-28. The two methods for making real-ear response determinations are comparable. Calculation of articulation index is shown in Figure A5-29 using the liquid pad and in Figure A5-30 using the foam filler. (They result in equal performance.)

5. David Clark Headset (Model 372-8C-M)

This headset has a Roanwell earphone which appears to be a modified H-79/AIC telephone receiver. Its frequency response at ground level is shown in Figure A5-31. No altitude response measurements were made as this headset is limited to ground application. The physical dimensions of the earphone are the same as the H-79/AIC earphone. Acoustical sealing is provided by a molded foam rubber pad cemented to the body of a plastic shell.

Real-ear response measurements made by both loudness balance and threshold techniques are shown in Figures A5-32 and A5-33. Sound attenuation provided by the headset is shown in Figure A5-34. The increase in attenuation provided by this headset over the H-158 headset is attributable to increased headband tension, the larger mass of the David Clark and the greater enclosed volume.

Articulation index calculations are shown in Figure A5-35.

APPENDIX 5.2

5.2 INSERT EARPHONES

In the initial consideration of insert earphones, noise attenuation for the Harvintip taken from Reference 11 was used and is shown in Figure A5-36. It was not planned to make additional measurements of the noise attenuation provided by various ear plugs. However, in the course of making measurements of the response of a bone conduction receiver with the ear plugs in place, the attenuation provided by the ear plug was measured in order to insure that it was inserted properly. Mine Safety Appliance Company Ear Defenders* were used and the real-ear attenuation at threshold was measured. The results are shown in Figure A5-37. It is seen that excellent attenuation is indicated. It should be noted, however, that initially the subjects were not able to insert the Ear Defenders in such a way as to obtain appreciable low frequency attenuation. When inserted by the wearer in such a way that both the wearer and experimenter thought the device was fitted properly, the observed attenuation was very small. After six to eight trials, and without any recognized difference in technique, the observed attenuation increased and consistent values of attenuation were obtained with subsequent insertions.

Assuming that an earphone can be coupled to the ear canal by means of a Harvintip or something similar to the MSA Ear Defender without loss of noise attenuation, we can make some estimate of the size, weight and response of a usable earphone from existing units. For example, Figure A5-38 shows the response in a 6cc coupler of an experimental moving-coil earphone, 3/8 inch thick, 7/8 inch in diameter and weighing 10 grams. Using this response and the attenuation of the MSA Ear Defenders, the articulation index was calculated in Figure A5-39 and was found to be .97 (assuming 200 mw and no peak clipping), giving substantially perfect intelligibility.

Actually, 2cc coupler response would have been more appropriate, so that the above results are no doubt conservative, if the attenuation can be consistently realized in practice. In addition, any increase in attenuation provided by a helmet would result in a corresponding decrease in sensitivity requirements for the earphone.

* The Mine Safety Appliance Ear Defenders are essentially similar to the V-51R Ear Warden.

APPENDIX 5.3

5.3 EARPHONE LOCATED IN OTHERWISE UNOCCUPIED SPACE INSIDE THE HELMET AND CONNECTED BY TUBES TO THE EAR

Placing earphones over the ears necessarily increases the overall dimensions of a pressure helmet. It has often been suggested that the earphones might be located in otherwise unoccupied space inside the helmet and the sound conducted to the ears by means of tubes. The purpose of this study was to determine , with a limited amount of experimentation, something of the feasibility of this approach.

As with other headset arrangements the principal factors determining communication performance are earphone response (including sensitivity) and noise attenuation. Noise attenuation may be limited by transmission through the connecting tubes or by the connection between the tube and the ear canal. The connection between the tube and the ear canal will certainly be very critical from the standpoint of both comfort and noise reduction. Consideration of this aspect of the problem was considered beyond the scope of the present investigation. The aim here was to determine whether the approach is sufficiently promising to warrant recommending that the necessary development of a seal to the ear canal be undertaken.

The approach taken in this phase was to connect an earphone to a 2cc coupler by various tubes, ranging from 2 to 8 inches in length and 0.135 to 0.375 inch in diameter. Both flexible and rigid tubes were used. The results serve to indicate:

1. problems involved in obtaining the desired frequency response and sensitivity
2. effect of altitude on response
3. noise attenuation attainable (neglecting seal to ear canal).

(1) Frequency Response

Frequency response characteristics of various earphone-tube arrangements were determined using an H-143/AIC dynamic earphone, 640-AA condenser microphone, and 2cc coupler as sketched in Figure A5-40. The earphone was driven by a beat-frequency oscillator and the output of the 640-AA microphone was measured using an automatic frequency response recording apparatus. The effect of altitude on frequency response was determined by raising the system to a simulated altitude of 25,000 feet in a vacuum chamber. The 2cc coupler response of the H-143/AIC earphone is shown in Figure A5-41.

Figure A5-42 shows the effect of varying the tube length while the diameter of the tube remains constant. Figure A5-43 shows the effect of varying diameter with constant 6-inch length. All tubes were transparent flexible vinyl sleeving of the type used for high voltage insulation in electric circuits. The shift in resonances with varying tube length is apparent. Changing the diameter affects primarily the lowest resonance, although some increase in attenuation is noted as the diameter is decreased. There is a trend for falling high frequency response, but this could probably be corrected, at least to some degree, by proper earphone design.

The effect of altitude on frequency response is shown in Figure A5-44. The data in Figure A5-44 was obtained using a 6-inch length of rubber tubing, 0.140 inch inside diameter. The effect of altitude was found to be negligible regardless of the size of the tube or the nature of the material.

(2) Noise Attenuation

The attenuation of external noises by various tubes was measured in the following manner. A 640-AA condenser microphone with a probe tube attachment was placed one meter from a loudspeaker in an anechoic chamber. The loudspeaker was driven by a beat-frequency oscillator and amplifier so as to produce a sound pressure of at least 100 db at the microphone at all frequencies between 100 and 8000 cps. Curves of the sound pressure vs frequency were drawn (a) with the probe tube open, and (b) with the tube in question cemented over the probe tube, the remote end of the tube being closed with a brass plug and rigidly clamped to the microphone support. The difference between the two curves was taken as the attenuation produced by the tube. To check on possible pick-up through the walls of the microphone itself, the end of the probe tube was sealed. The sound pressure recorded was at least 10 db below that pressure recorded with a "sample" tube in place (indicating negligible error due to transmission through the microphone or probe).

Figure A5-45 shows the noise attenuation measured for various tubes. Sample A is a rigid phenolic tube of the type used for coil forms and other voltage-insulating applications. Sample B was formed by drilling out a length of flexible teflon rod. Sample C was constructed from a close wound steel spring covered with flexible vinyl tubing. Sample D was a length of rubber tubing of the type found in a chemical laboratory. Sample E and F were vinyl tubing; Sample E being transparent and Sample F having a fabric base.

It can be seen that the greatest noise attenuation is obtained from a rigid tube. When a brass tube was tested the attenuation was so great that the acoustical noise picked up by the microphone masked the noise level of the system. The best of the flexible tubes was the most nearly rigid, the teflon tube. However, even the flexible tubing provides substantial attenuation in the speech range.

(3) **Evaluation of Commercially Available Units**

Two headsets which use tubes to transmit sound to the ears were obtained and evaluated.

The Telex "dynaset" uses a dynamic transducer of nominal 6 ohm impedance. Sound is transmitted to the ears by two metal tubes, 0.150 inch inside diameter by 8 inches long. The 2cc coupler response of the Dynaset is shown in Figure A5-46 The real ear response at threshold was determined by the method described in Appendix F and is also shown in Figure A5-46 It is seen that there is a substantial difference in the two response curves.

Some investigation was made of this discrepancy and it was found that as the input power was increased there was a sudden increase in the 2cc coupler response of the transducer of approximately 20 db. Attempts were made to explore this phenomenon more fully, but was limited because of time and funds. It was found that there was a change in response as a function of input power. This change was non-linear, however a definite relationship could not be established. The greatest change in response occurred around 1 milliwatt of input power. Therefore, it is felt that the real-ear response data as measured at threshold is somewhat questionable and most probably on the low side. It is possible that real-ear response measurements by loudness balance techniques would be more reliable and would be comparable to the 2cc coupler response curve shown in Figure A5-46.

The Telex "Earset" also uses a 6 ohm dynamic transducer but sound is conducted to only one ear by a 5 foot length of vinyl tubing 0.150 inch inside diameter. The 2cc coupler response of the earset is shown in Figure A5-47. Of interest is the damping effect of the vinyl tubing on the resonant peak in the transducer response. Real-ear response measurements were not made.

Noise attenuation measurements were not made since these headsets are designed for use in low noise levels and are loosely coupled to the wearer's ear.

Figure A5-48 shows the articulation index calculated using the 2cc coupler response of the H-143 (Figure A5-44) achieved with the six inch rubber tube, 0.140 inch diameter and simulating coupling to the ear via an MSA ear defender. It is seen that almost perfect intelligibility is achieved with the combination.

Figure A5-49 shows the articulation index calculated using the 2cc coupler response of the Telex "Earset" coupler to the ear via an MSA ear defender. This combination also results in almost perfect intelligibility.

Figures A5-50, A5-51 and A5-52 show articulation calculation of various combinations of earphones and couplers. The calculated performance of any of these combinations is comparable to and, for the most part, better than the AN/AIC-10 system.

APPENDIX 5.4

5.4 REMOTE EARPHONE

This phase of the study involved the exploration of the possibility of using a transducer mounted in some unoccupied space in a helmet.

The work done at Armour (References 13 and 14) establishes the sound attenuation which can be provided by a helmet (Figures A5-53 and A5-54). The latter figure was subjectively determined using two subjects - one trial each subject. The real-ear response of a transducer "loosely" coupled to the ears was ascertained in Reference 12 and is shown in Figure A5-55. In this case an H-79/AIC (high sensitivity) earphone was used in a pressure helmet (MA-1) and coupled to the ears by a rather poorly fitting polyurethane helmet liner. The earphones were located approximately over the ears.

Utilizing this information the articulation index was calculated as shown in Figure A5-56 resulting in a word articulation of 43%. It is necessary to point out at this time that the calculation was based on 200 mw of available power; if, however, the power were increased to 2 watts the word articulation would be appreciably increased to about 80%.

Additional investigation carried out in this laboratory was to ascertain the response as measured with a probe tube when the transducer is located near the nape of the neck. An experimental pressure helmet shell was placed on a manikin and sealed around the neck with clay as shown in Figure A5-57. The manikin had a microphone at the ear position as shown. The volume enclosed by the helmet was approximately 3000cc. Measurements were made using an H-143 earphone and also an RCA 2-1/8" Inverted Pot magnet structure speaker Type No. 239S-1. The response obtained with these units is shown in Figures A5-58 and A5-59. Word articulation calculations (Figures A5-60 and A5-61) were made using the sound attenuation data previously discussed. Location of the transducers at the nape of the neck (without auxiliary means for conducting the sound to the ears) is not as effective as a transducer located near the ears (assuming that power is limited to 200 milliwatts).

However, if increased power is considered, we find that about 2 watts would result in an articulation index of .71. In addition, it should be possible to provide channels in the helmet liner which would conduct the sound to the ears more efficiently than in the arrangement in Figure A5-57.

Figure A5-62 is also of interest in connection with this approach. It shows the response obtained with an open helmet (similar to an infantry helmet) and with one Dyna Labs insert earphone driving two tubes which lead to a horn-like arrangement at the ear,

but do not seal over the ear or make any contact with the pinna. A diagram of the arrangement is shown in Figure A5-63. It is seen that the sensitivity (Figure A5-62) is quite high over a reasonably wide band. This leads one to believe that the general approach is technically feasible and might even be accomplished with presently available power if an efficient channeling arrangement can be incorporated in the helmet liner.

APPENDIX 5.5

5.5 BONE CONDUCTION

Two different arrangements were explored; a bone conduction receiver located on the forehead and an experimental earphone located on the forehead and coupled by means of its earcushion. The forehead location was used, rather than the usual location over the mastoid, because of related studies in process at W.E.A.L. in regard to speech pickup from the forehead.

The bone conduction receiver used was a Western Electric Company "Audiophone", a commercial hearing aid receiver. A bone conductor might be used in conjunction with ear plugs to provide noise attenuation or with noise attenuation provided by a helmet. It was desired to obtain the response of the bone conduction unit with the ears occluded and with the ears essentially open. Reference 15 indicated that there could be up to 24 db difference in thresholds for the two conditions. However, there was some question whether the subject was hearing a bone conducted signal or an air conducted signal since the bone conduction receiver radiates some sound into the air. In order to eliminate the possibility of his hearing an air conducted signal, some measurements were made with the bone conduction receiver covered with a muff-type ear protector (RCA "Quiet-Ear"). No significant difference was observed. Thresholds were then determined for two conditions: (1) MSA Ear Defenders in both ears, and (2) ears covered by the "Quiet-Ear" ear protector. The open ear condition was not measured because of the question of air-conducted sound.

The response of the bone conduction receiver was measured at threshold in the following manner. The Bekesy Audiometer was used to determine the subject's threshold in a free-field. The output from the audiometer was then applied to the bone conduction unit and the subject's threshold again determined. The voltage applied to the bone conduction unit during this determination was measured. The equivalent free field pressure for constant voltage (1 volt) across the bone conduction receiver was determined from these two threshold determinations. The impedance of the bone conduction receiver was measured and a final response curve calculated assuming a resistive generator matching the impedance of the bone conduction receiver at 1000 cps with one milliwatt of power available. These response curves are shown in Figure A5-64.

The response of one earphone of headset H-158(XA)/AIC placed on the forehead was determined similarly, except that in this case the impedance was constant and resistive so that the correction for impedance was not necessary. The response of this arrangement is shown in Figure A5-65. Comparison with the real-ear response of this headset shown in Figure A5-28 shows a decrease in response of roughly 65 db.

Thus approximately 4×10^6 times as much power would be required for this arrangement than would be required for the conventional use of headset H-158(XA)/AIC. Or, for the same amount of power, an additional 65 db of noise attenuation would be required.

APPENDIX 5.6

5.6 SUBJECTIVE MEASUREMENTS

For some items for which no data existed, subjective measurements were made of real-ear response and real-ear attenuation at threshold. These tests followed standard techniques and procedures; however, for the sake of clarity they will be briefly explained.

Nine male Rutgers University students were used as subjects. They were physically checked by an ear specialist and were found to have normal hearing. Later threshold tests confirmed their normality. The students were given preliminary instructions in the use of all the devices tested. For expediency most of the measurements were made on five of the subjects; tests were repeated on each subject on three separate occasions. The average of fifteen separate determinations and the standard deviation at each test tone was calculated and is shown in the various curves associated with a particular item.

I. Real-Ear Response - Loudness Balance

In this measurement a sound source and a listener position are situated far enough apart in a free field so that the sound wave arriving at the listener position approximates a free, progressive plane wave. The source was calibrated to provide a uniform sound-pressure level over a wide frequency range at the listener position. The sound pressure level was maintained at 90 db. A block diagram showing the various components required for this test is shown in Figure A5-66.

A listener or subject occupies the listening position facing the sound source along its principal axis and listens to the sound wave at some specified test frequency. Then the source tone is stopped and the listener puts the headset on, adjusting it to fit in a typical manner with regard to reasonable comfort. An earphone tone of the same frequency is then presented to the subject. The variable attenuator is adjusted until the loudness of the earphone tone is judged to be equal to that of the source (reference) tone. The earphone tone is stopped, the headset removed and the reference tone again sounded for comparison. This is repeated a number of times until the listener arrives at an attenuator setting which, in his judgement makes the two tones equal in loudness.

An electronic keying device was used to alternate the test tone between the loudspeaker and the headset under test.

When the subject has judged the tones from both sources to be of equal loudness he signals the experimenter who then records the terminal voltage across the headset. Several judgements are made at each frequency. Taking the average of different determinations by different subjects, the experimenter obtains the real-ear response.

II. Measurements at Threshold

Measurements at threshold were done with a Bekesy Audiometer, Grayson-Stadler Model E-800, in a small, highly damped room. A comparison of the thresholds measured with the test crew with the Minimum Audible Field established by Sivian and White is given in Figure A5-67.

The audiometer consists of an oscillator and an automatic recorder. The horizontal drive on the chart is coupled to the oscillator dial. The range from 100 to 10,000 cps can be scanned in 7 or 3.5 minutes, depending on the drive speed.

The recording pen is coupled to a logarithmic attenuator which controls the signal level at the output of the audiometer. The recording pen and attenuator are driven by a motor which can be reversed by a switch under the control of the subject. The pen motor normally operates to increase the signal to the subject. When he hears the signal he presses the switch, reversing the motor attenuating the signal until he no longer hears it. The signal level thus crosses and recrosses the subject threshold of hearing as the frequency is slowly changed. The rate of change of signal level is 140 db/minute with the slow chart speed and 280db/minute with the fast chart speed.

The range between decision that the signal is present and that it is not present is of the order of 10 - 15 db. The actual threshold is taken as the mean of the extreme deviations as plotted in the audiometer chart. A typical chart is shown in Figure A5-68.

A. Real-Ear Attenuation at Threshold

As far as practicable, these tests were conducted in conformance with the proposed American Standard for the measurement of the Real-Ear Attenuation of Ear Protectors at Threshold, Z 24.22/406.

In this method, the subject is seated facing a loudspeaker in the test chamber. He is presented with a tone from the audiometer, pulsed four times per second, which he adjusts to his threshold as described above. Having made threshold determinations from 100 to 10,000 cps in this manner, he then dons the headset or attenuating device. He is then presented with a white noise signal and he adjusts the headset for minimum sound transmission. He is then presented with the pulsed tone and repeats the threshold determination as before. The difference between the two threshold curves plotted on the audiometer is then taken as the noise attenuation of the device.

B. Real-Ear Response by Threshold Comparison

In this method, the subject first establishes his free-field threshold curve using the signal from the loudspeaker in the chamber. He then dons the headset device and adjusts it for proper fit. The audiometer output is then fed into the headset and the subject establishes a new threshold curve. The audiometer plot then corresponds to the voltage on the speaker and headset which produce signals at the observer's threshold.

The difference of the two threshold curves is then corrected for the free field calibration of the speaker as measured with a W.E. 640-AA microphone at the subject's head position (subject out of the room). The result is taken as the real ear response by the threshold comparison method.

APPENDIX 5.7

5.7 CALCULATION OF ARTICULATION INDEX

A. Calculation For A Specific Earphone-Coupler Combination

In this analysis the method of articulation index calculation, developed by French and Steinberg (Reference 4) and by Beranek (Reference 5) is applied. This method is based on experimental data which show that the dynamic range of speech (considering r.m.s. pressures in 1/8 second intervals) is 30 db, and the frequency range which contributes to intelligibility is 200 to 6100 cps. Therefore, a communication system which provides a signal-to-noise ratio (rms speech peaks to rms noise) at the listener of 30 db or better over the entire 200 to 6100 cps band will provide "perfect" articulation performance. The method provides for calculation of word articulation scores as follows:

(1) The frequency band from 200 to 6100 cps is divided into 20 bands which contribute equally to intelligibility. Each band can thus contribute a maximum of 5% to the word articulation performance.

(2) The effective dynamic range of the speech signal at the listener's eardrum in each band is then determined. According to Reference 6, best correlation with experiment is obtained when the dynamic range at the listener is limited at high sound pressures to a spectrum level of 95 db/cycle. The dynamic range is limited at low sound pressures by the listener's threshold of hearing or by noise which raises the listener's effective threshold of hearing, that is, masks the speech signal. A dynamic range greater than 30 db in any band contributes no more to intelligibility than does a 30 db range.

(3) A summation is then made over all bands of the product of the fraction of the maximum usable dynamic range in each band and the maximum possible contribution of each band. The result is called the articulation index, abbreviated A.I.

(4) The next step is to determine the relation between articulation index and word articulation scores (W.A.) for the particular crew of talkers and listeners and the particular material transmitted. When this latter relation has been determined the word articulation score for various conditions can be calculated, and while it holds strictly only for the test crew used, it applies approximately to the population in general, the degree of approximation depending upon the adequacy of the test crew as a sample of the population.

The relation between W.A. and A.I. given in the paper by Beranek (Reference 5) is shown as Curve B of Figure A569. Later experience at RCA has indicated that for PB word lists and crews with moderate training, Curve B of Figure A569 gives optimistic

results. Curve M (See Reference 6) of Figure A5-69 shows the relation determined for an RCA test crew and shows good agreement with a similar relation reported by Pollack (Reference 7) shown as Curve P of Figure A5-69. Curve M will be used for estimates of word articulation scores. No consideration was given to the microphone problem in this report. For the purpose of making articulation calculations the shape of the speech spectrum was as shown in Figure A5-70 which was based upon Reference 5.

With reference to the methods suggested in References 4 and 5, an articulation index scale was used. The frequency scale is distorted to make equal increments give equal contributions to articulation index.

To obtain the spectrum level of sound pressure at the eardrum it is necessary to consider the amplifier capabilities and the departures from uniform orthotelephonic response from the earphone-earcushion combination as positioned on a subject's head. For example: If we have a headset which has a real ear response as in Figure A5-71, then the spectrum level of speech peaks in equivalent free field sound pressure for a 200 milliwatt amplifier will appear as the lower curve in Figure A5-72 assuming uniform microphone and amplifier response. The latter curve is then plotted on articulation index computation chart - Curve A of Figure A5-73.

In addition to the above, it is necessary to know the amount of noise attenuation provided by the headset. The subjective method employed in obtaining real ear attenuation values was that of threshold shift. The 120 db jet noise spectrum is then reduced by the measured hearphone attenuation, shown in Figure A5-75 (The noise spectrum of Figure A5-74 was used by RCA and the Air Force in the development and testing of AN/AIC-10 equipment. It is used here only for the sample A.I. calculation in this appendix. For all other A.I. calculations in this report the noise spectrum of Figure A5-6 was used.)

The area between the spectrum level of speech peaks and the noise passing through the earcushions, is a measure of articulation index. By calculating the articulation index and then referring to Curve M of Figure A5-69 an estimate of word articulation scores can be made.

B. Calculation Of S.P.L. Required To Achieve An A.I. of 0.71

While it was originally intended to make the articulation index calculations on the basis of 200 milliwatts input power since this is the power available from AN/AIC-10 amplifiers, W.E.A.L. requested a somewhat more general approach, namely, that the overall sound pressure level necessary to produce an articulation index equivalent to that of present AN/AIC-10 headsets (A.I. = .71) be determined.

The procedure in this calculation was first to determine the transmitted noise spectrum by subtracting the amount of noise attenuation provided by the arrangement under consideration from the 120 db noise spectrum. Uniform orthotelephonic system

response was then assumed and the spectrum of speech peaks necessary to give an articulation index of 0.71 was established. The overall level of this speech peak spectrum was then determined and this value was placed on the chart. It can be used in conjunction with the 1000 cps sensitivity figure to estimate the power required; that is, subtracting the 1000 cps sensitivity in db from the S.P.L. in db for 0.71 A.I. gives the required power in db re one milliwatt.

The following list summarizes the results for the various arrangements considered.

Title	Figure No.	Spectrum Level Of Speech Peaks Having An Overall Level Of - (DB)
ANB-H-1/MC-162A	A5-76	122
ANB-H-1/5-B, 6	A5-77	117
ANB-H-1A/MC-162A	A5-78	112
ANB-H-1A/5-B, 6	A5-79	117
H-78B/AIC	A5-80	112
H-158/Liquid Pad	A5-81	104
H-158/Iso. Foam	A5-82	104
David Clark - 372-8C-M	A5-83	102
*MSA Ear Defender (V-51R)	A5-84	106
Harvintip	A5-85	111
Experimental Plastic Helmet	A5-86	108

* V-51R Ear Warden and Mine Safety Appliance Ear Defender are essentially similar.

REFERENCES

APPENDIX 5

1. HS-33 and HS-28 Headset, Memorandum Report TSERR2B5-169. Army Air Forces, Air Technical Service Command, Wright Field, Dayton, Ohio, (26 October 1945).

2. Tests at ANB-H-1A Earphones (Permoflux Corporation Type PDR-3), Memorandum Report MCREE-48-20, U.S. Air Force, Air Materiel Command, Wright-Patterson Air Force Base, Dayton, Ohio, (3 May 1948).

3. "The Articulation Efficiency of Magnetic and Dynamic Earphones Used With Various Earphone Cushions in Noise," OSRD No. 1491, Psycho-Acoustic Laboratory, Harvard University, Cambridge, Massachusetts, (15 June 1943).

4. French and Steinberg, "Factors Governing the Intelligibility of Speech Sounds," JASA, Vol. 19, Page 90, (1947).

5. L. L. Beranek, "Design of Speech Communication Systems," Proc. I.R.E. Vol. 35, Page 880, (1947).

6. D. W. Martin, "Uniform Speech-Peak Clipping in a Uniform Signal-To-Noise Spectrum Ratio," JASA, Vol. 22, Page 614, (September 1950).

7. I. Pollack, "Effects of High Pass and Low Pass Filtering on the Intelligibility of Speech in Noise," JASA, Vol. 20, Page 259, (May 1948).

8. AN/AIC-10(XA) High Intelligibility Interphone Equipment, Final Report, Appendix 2, Radio Corporation of America, Camden, New Jersey (Contract W-33-038-ac-18181).

9. D. W. Martin and M. L. Touger, "Loudness Balance Method For Earphone Response Measurements," JASA, Vol. 22, Page 1833-1837 (1950).

10. Product Improvement of AIC-10 Acoustical Components, Final Report, "Headset-Microphone H-157(XA)/AIC," Radio Corporation of America, Camden, New Jersey (Contract AF 33(600)-33172).

11. "The development of Insert Tips For Use With Miniature Earphones," W. A. Shaw and D. E. Yates, OSRD Report No. 6179 (October 1945).

12. "Some Acoustic Performance Characteristics of the Bill Jack MA-1 Helmet," Contract AF 33(616)-456, Radio Corporation of America, Camden, New Jersey (March 1956).

13. "Communication in Jet Noise Fields," Progress Report No. 3, Armour Research Foundation, Illinois Institute of Technology, Chicago, Illinois (July 1957).

14. Ibid., Progress Report No. 6 (August 1957).

15. Watson, N. A. and Gales R. S., "Bone Conduction Threshold Measurements: Effects of Occlusion Enclosures and Masking Devices," JASA., (1943,) Vol. 14, pp 209-215.

16. WADC Technical Note 55-355 "Criteria for Short Time Exposure of Personnel to High Intensity Jet Aircraft Noise", (September 1955).

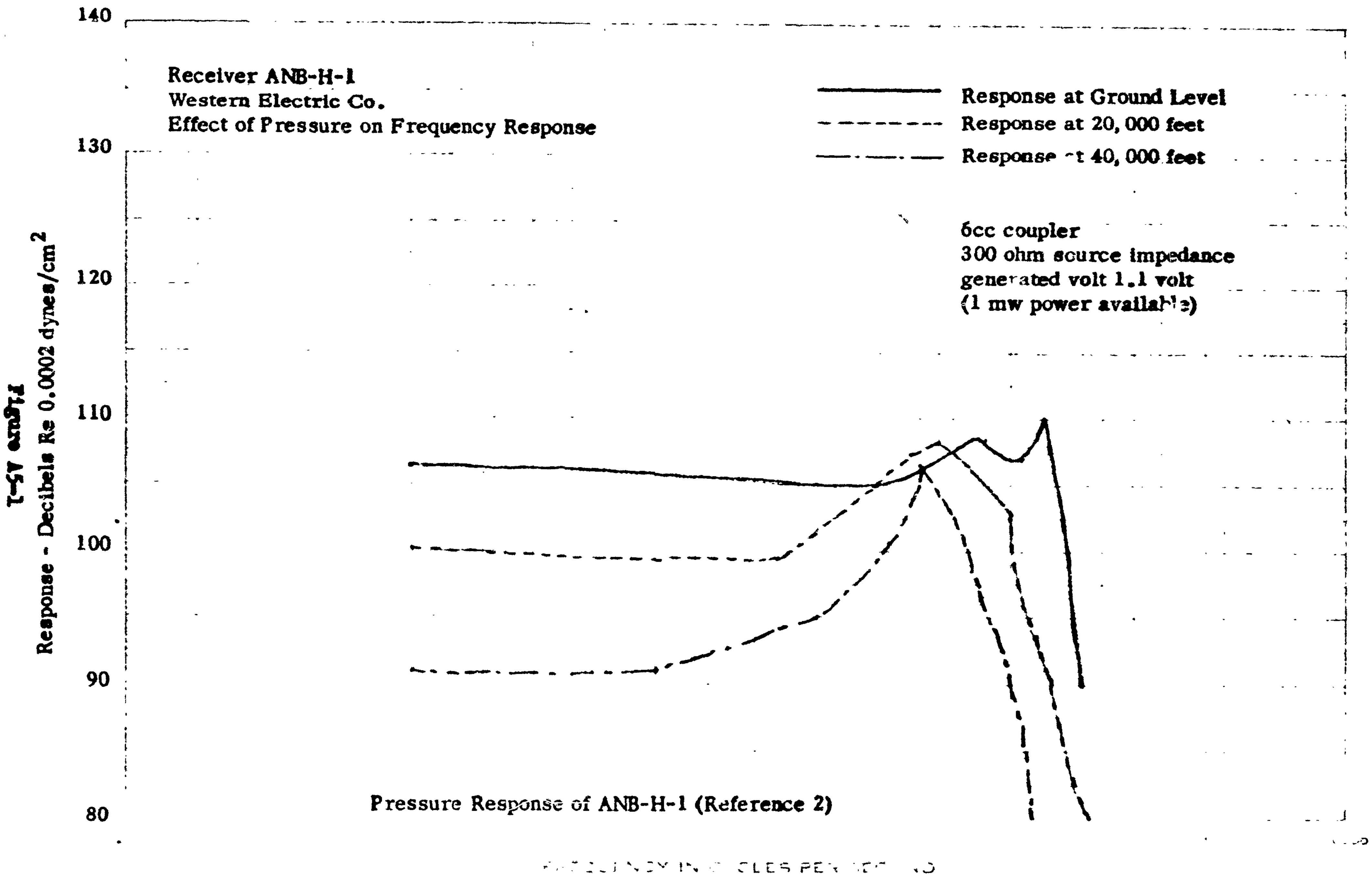
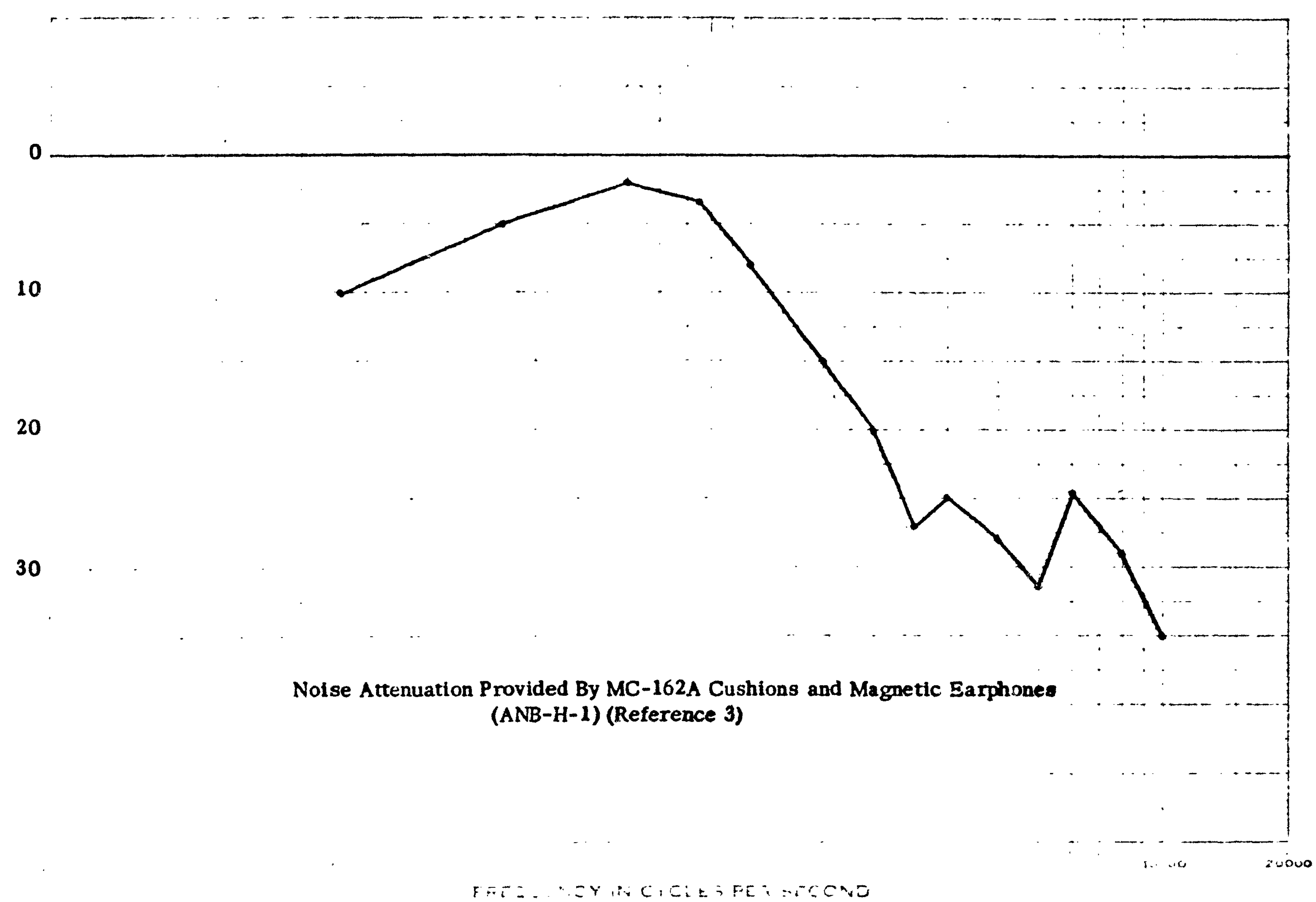

140
130
120
110
100
90
80
Response - Decibels Re 0.0002 dynes/cm²
Receiver ANB-H-1
Western Electric Co.
Effect of Pressure on Frequency Response
Response at Ground Level
Response at 20, 000 feet
Response at 40, 000 feet
6cc coupler
300 ohm source impedance
generated volt 1.1 volt
(1 mw power available)
Pressure Response of ANB-H-1 (Reference 2)

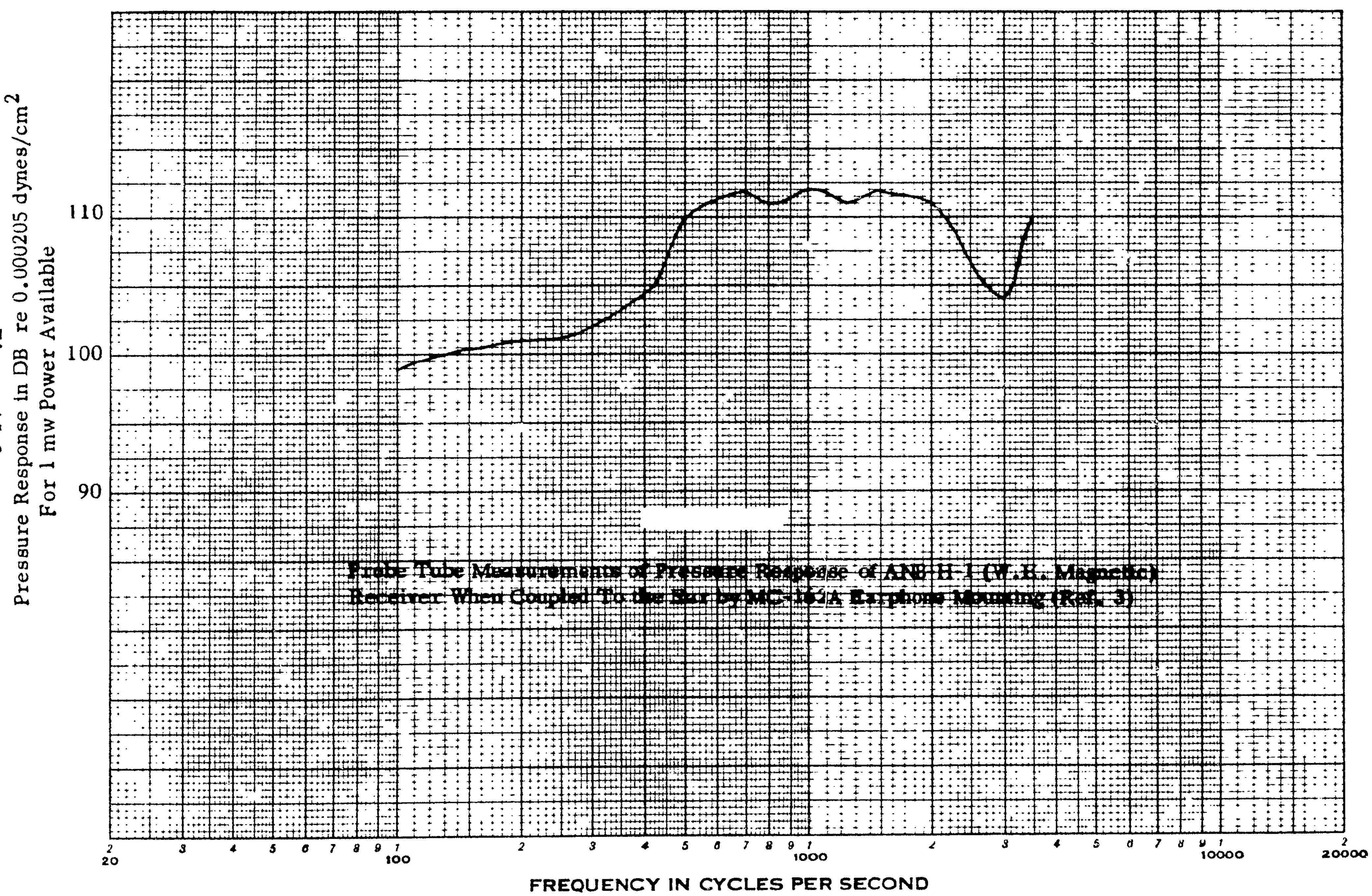
Figure A5-1

Figure A5-2



Noise Attenuation Provided By MC-162A Cushions and Magnetic Earphones
(ANB-H-1) (Reference 3)

Figure A5-3


Pressure Response in DB re 0.000205 dynes/cm2
For 1 mw Power Available
110
100
90
Probe Tube Measurements of Pressure Response of ANB-H-1 (W.E. Magnetic)
Receiver When Coupled To the Ear by MC-162A Earphone Mounting (Ref. 3)
20
100
1000
10000
20000
FREQUENCY IN CYCLES PER SECOND

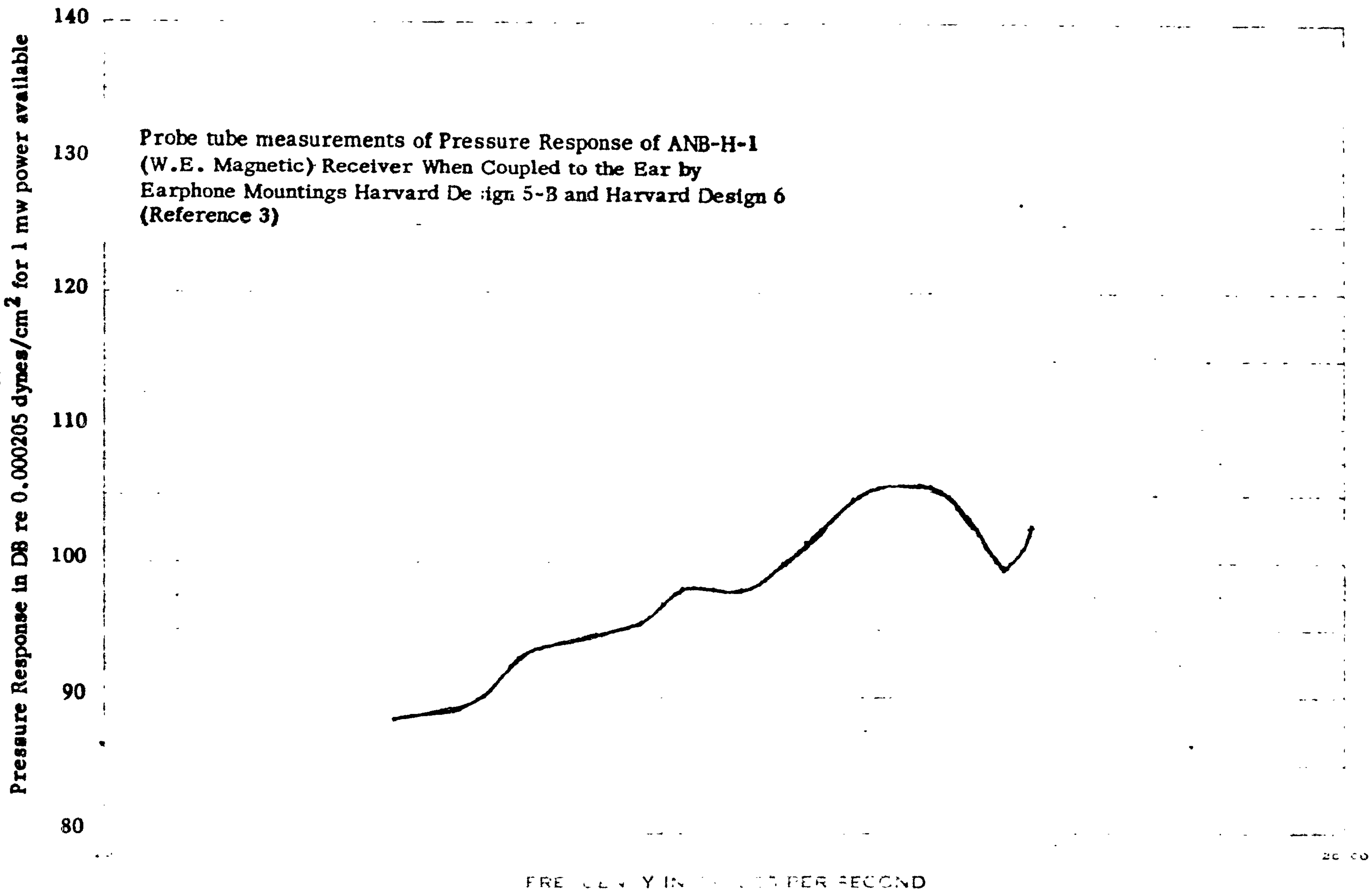

Probe tube measurements of Pressure Response of ANB-H-1
(W.E. Magnetic) Receiver When Coupled to the Ear by
Earphone Mountings Harvard Design 5-B and Harvard Design 6
(Reference 3)
140
130
120
110
100
90
80
Pressure Response in DB re 0.000205 dynes/cm² for 1 mw power available
PER SECOND

Figure 15-4

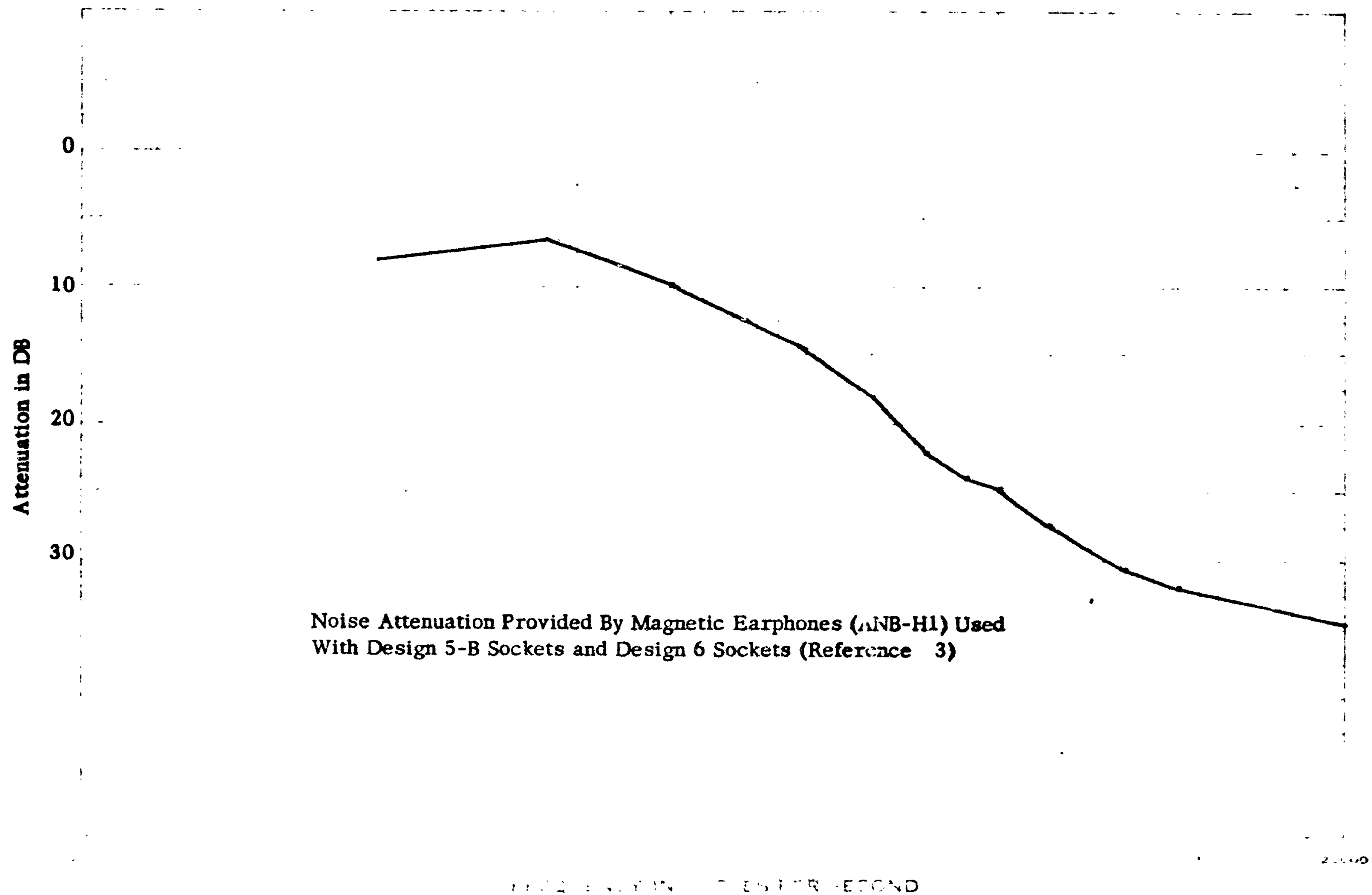

0
10
20
30
Attenuation in DB
Noise Attenuation Provided By Magnetic Earphones (ANB-H1) Used
With Design 5-B Sockets and Design 6 Sockets (Reference 3)

Figure A5-5

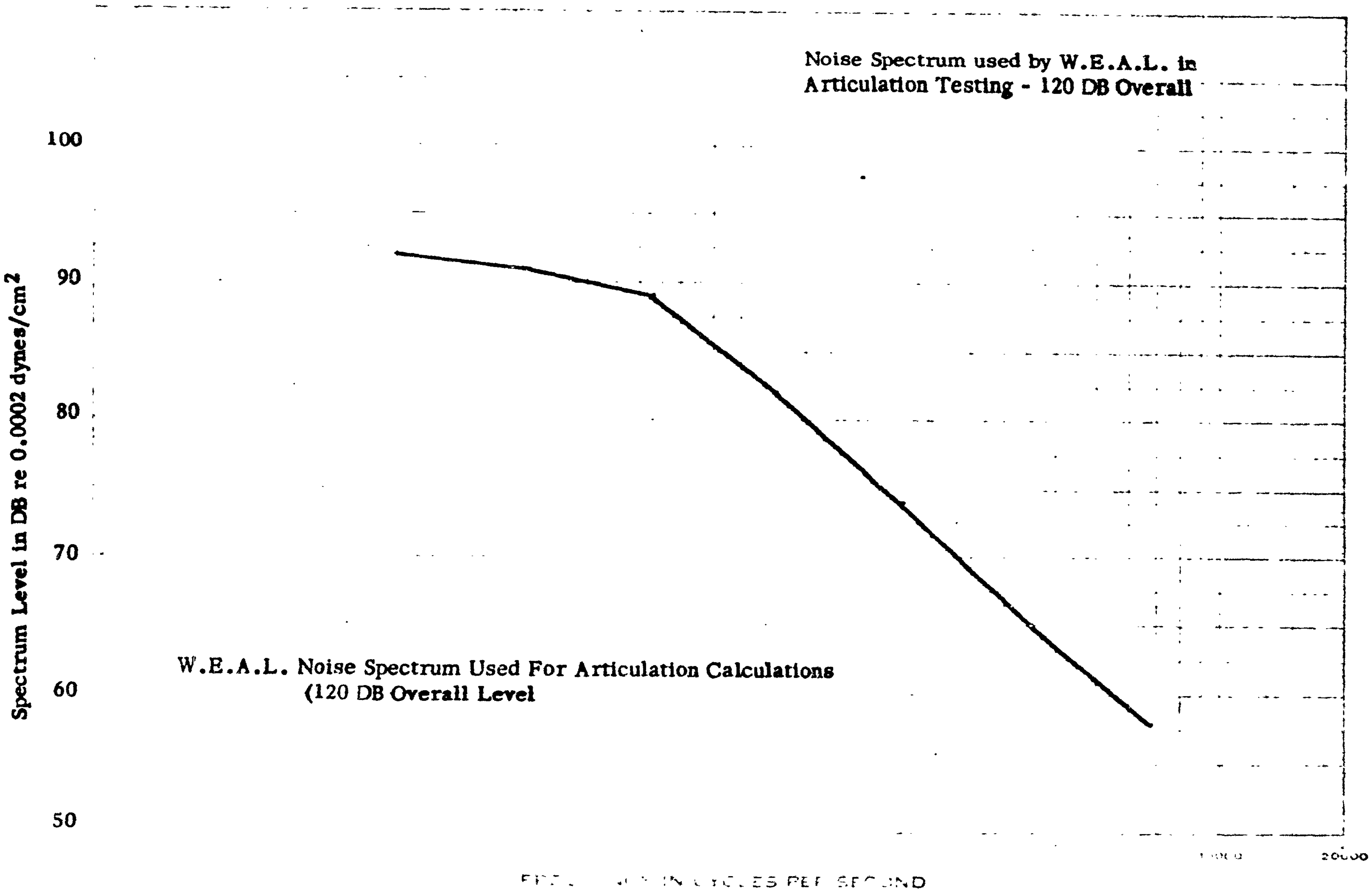
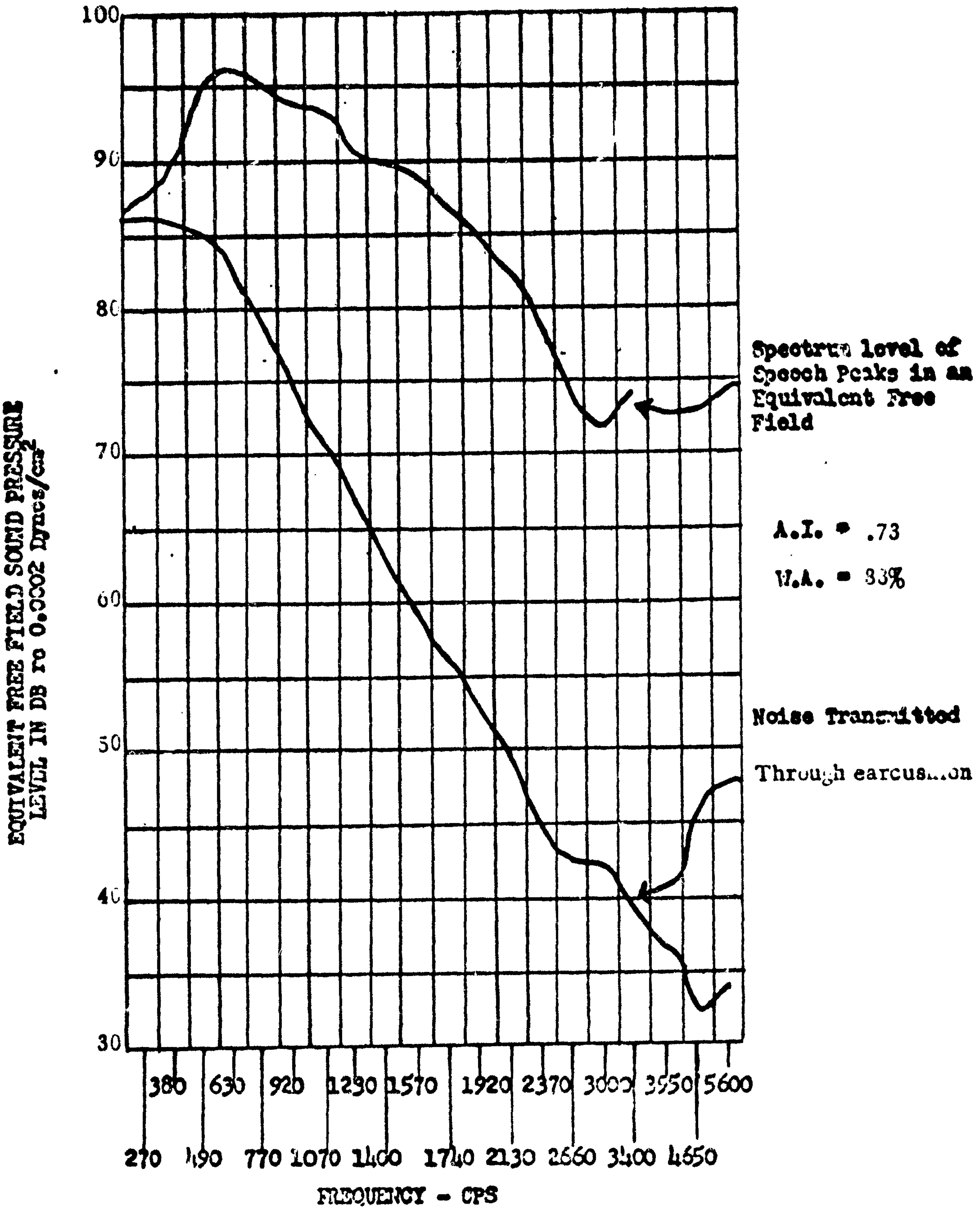

Noise Spectrum used by W.E.A.L. in
Articulation Testing - 120 DB Overall
W.E.A.L. Noise Spectrum Used For Articulation Calculations
(120 DB Overall Level
100
90
80
70
60
50
Spectrum Level in DB re 0.0002 dynes/cm2
20000

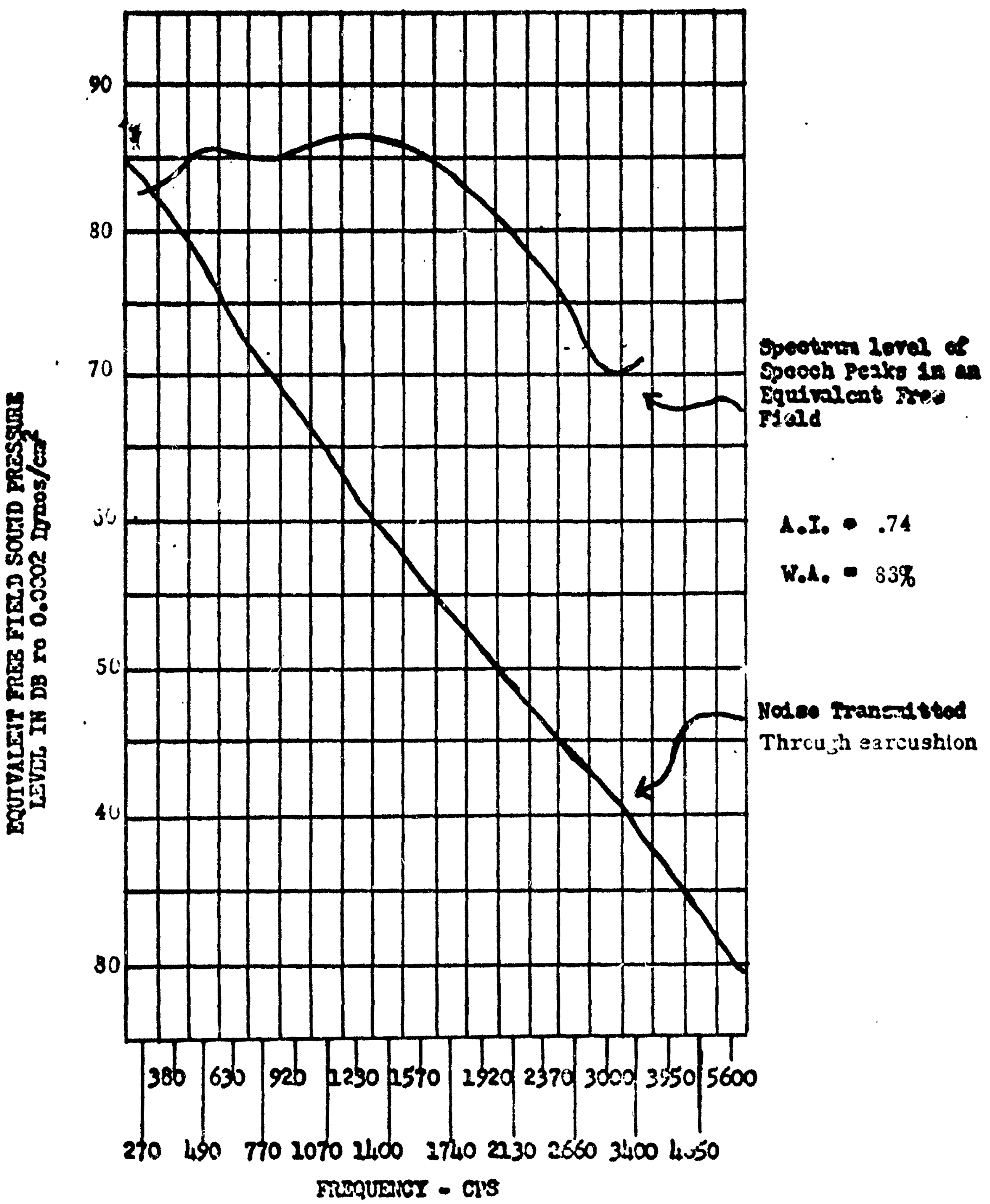
Figure A5-6



ARTICULATION INDEX COMPUTATION CHART
FOR

ANB-H-1 MC-162A Earcushion 200 mw
No Clipping

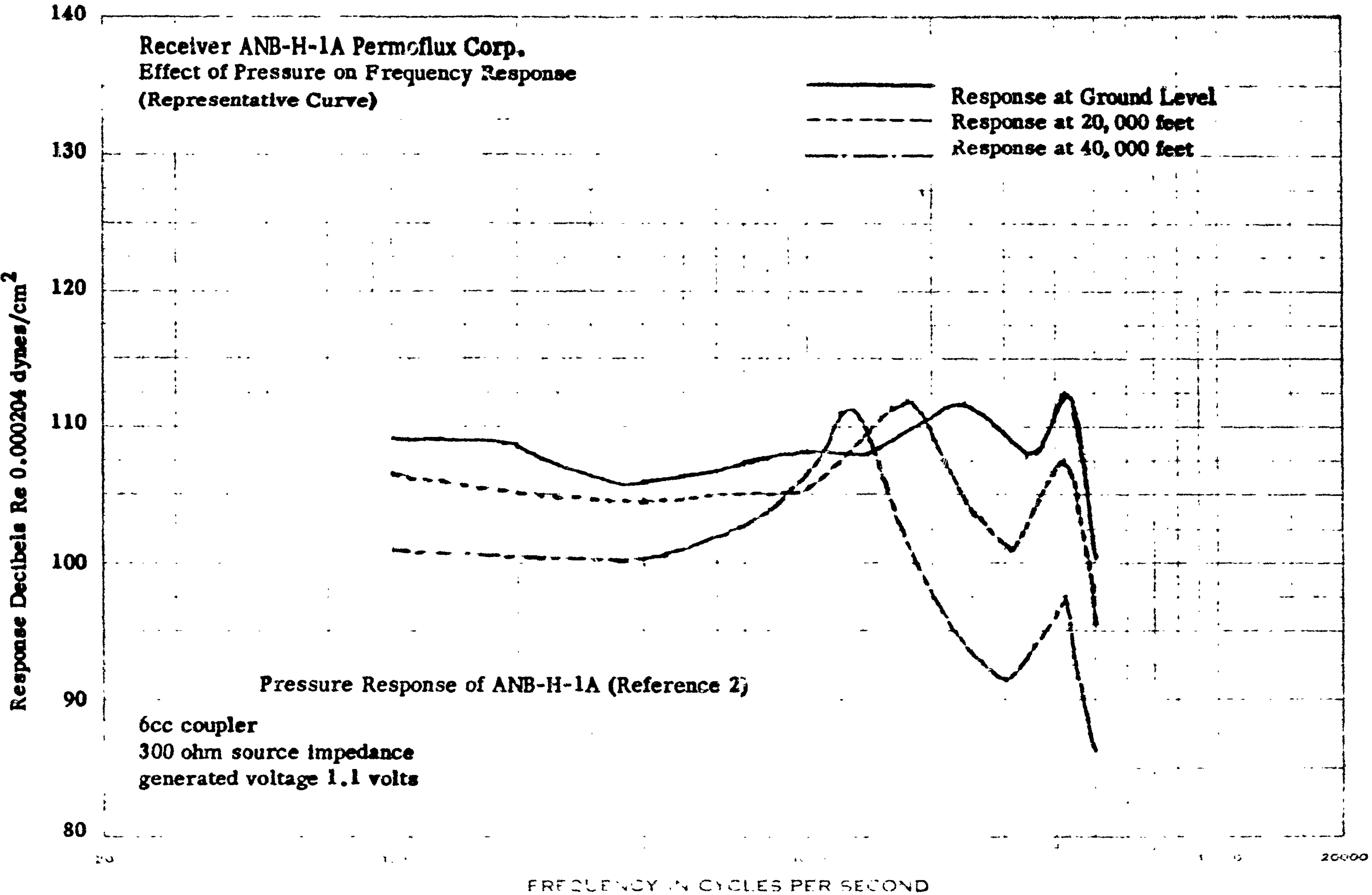
Figure A5-7



ARTICULATION INDEX COMPUTATION CHART FOR

ANB-H-1 Harvard Design Earcushions 5-B, 6
(200 mw, no clipping)

Figure A5-8



Receiver ANB-H-1A Permoflux Corp.
Effect of Pressure on Frequency Response
(Representative Curve)
Response at Ground Level
Response at 20,000 feet
Response at 40,000 feet
Pressure Response of ANB-H-1A (Reference 2)
6cc coupler
300 ohm source impedance
generated voltage 1.1 volts
140
130
120
110
100
90
80
20000
Response Decibels Re 0.000204 dynes/cm²
FREQUENCY IN CYCLES PER SECOND

Figure A5-9

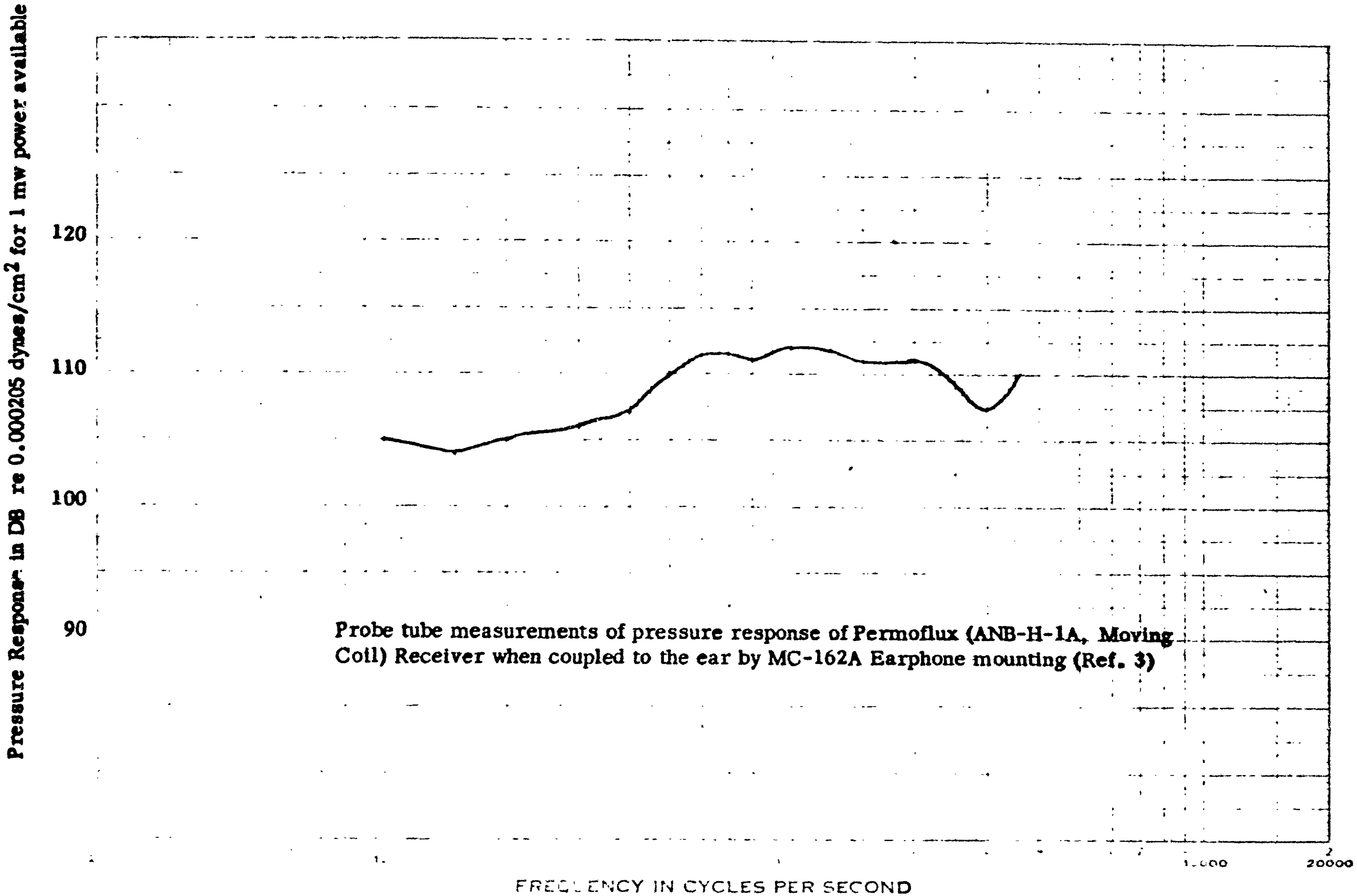

Pressure Response in DB re 0.000205 dynes/cm² for 1 mw power available
120
110
100
90
Probe tube measurements of pressure response of Permoflux (ANB-H-1A, Moving Coil) Receiver when coupled to the ear by MC-162A Earphone mounting (Ref. 3)
20000
FREQUENCY IN CYCLES PER SECOND

Figure A5-10

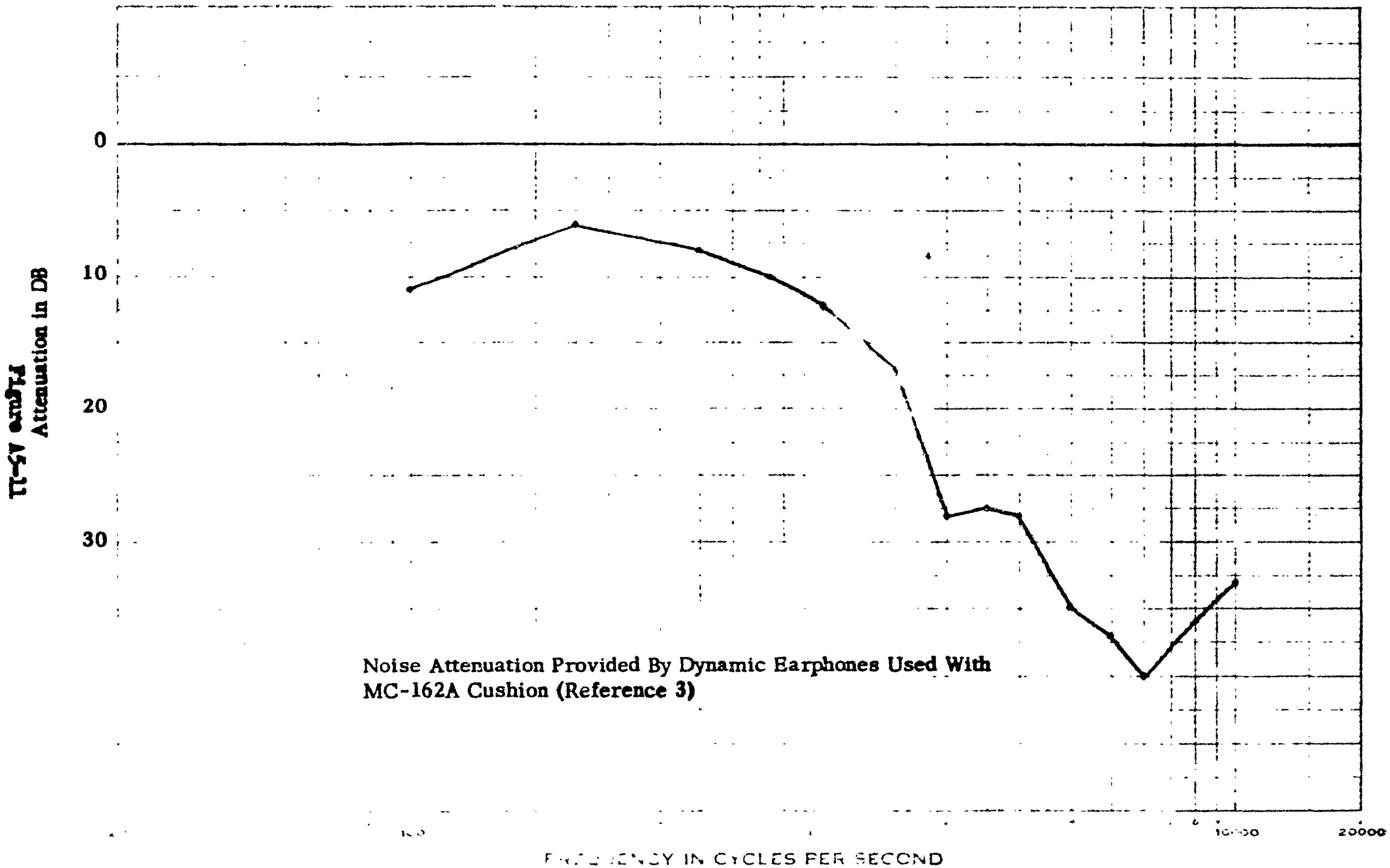
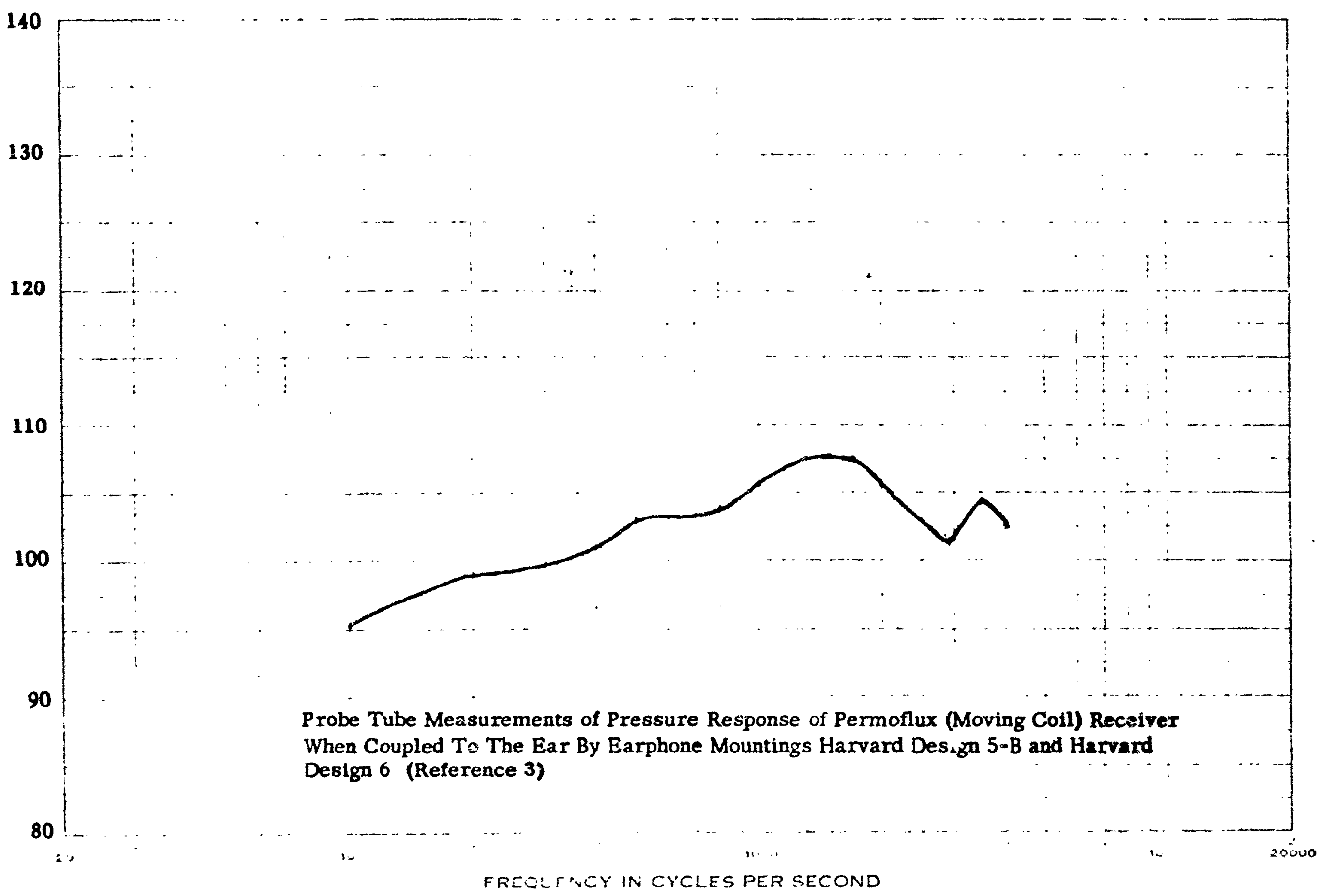
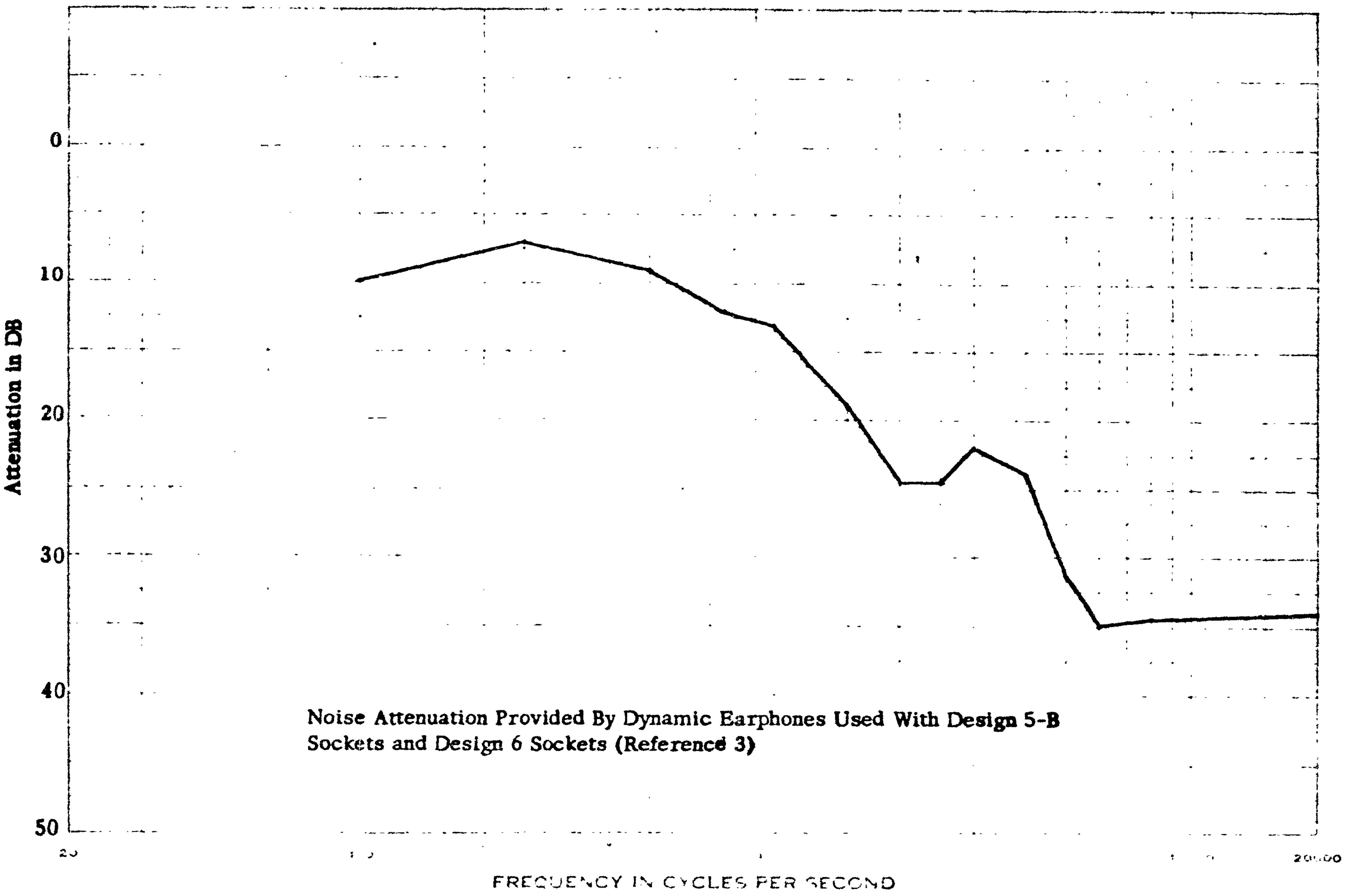

0
10
20
30
Attenuation in DB
Noise Attenuation Provided By Dynamic Earphones Used With
MC-162A Cushion (Reference 3)
20000
FREQUENCY IN CYCLES PER SECOND

Figure A5-11



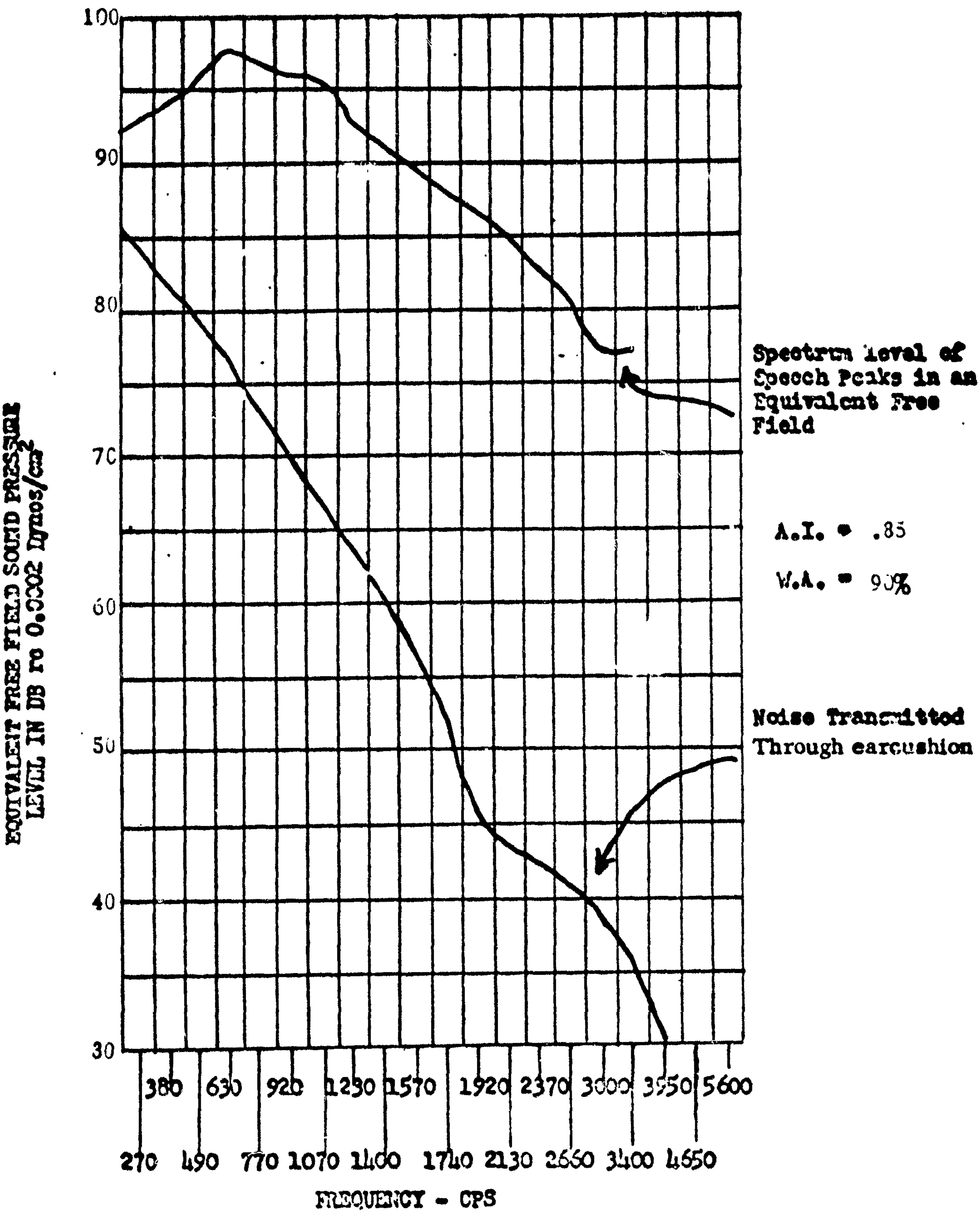
Pressure Response in DB re 0.000205 dynes/cm^2 per 1 mw

Figure A5-12



Noise Attenuation Provided By Dynamic Earphones Used With Design 5-B Sockets and Design 6 Sockets (Reference 3)

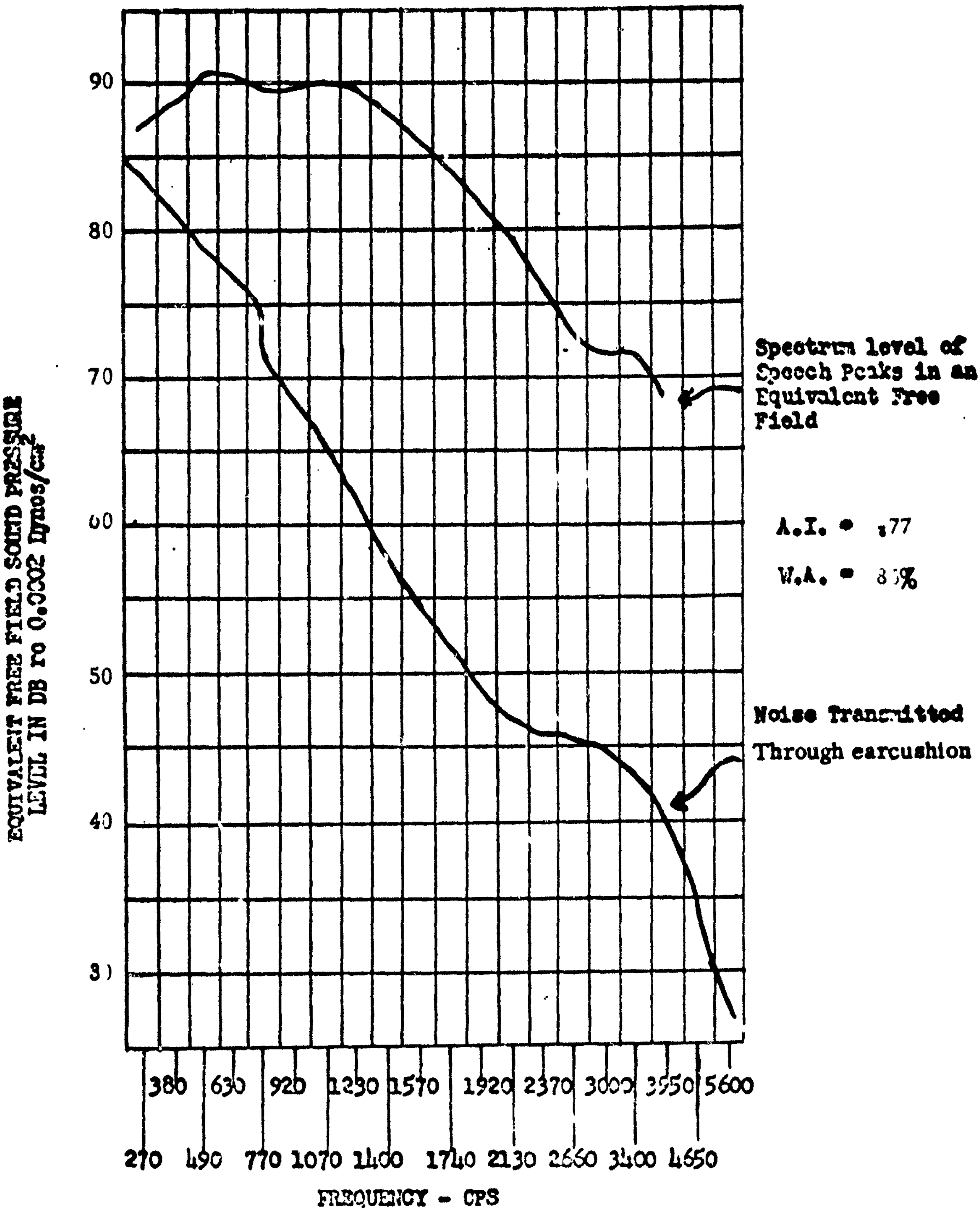
Figure A5-13



ARTICULATION INDEX COMPUTATION CHART FOR

ANB-H-1A Earcushion MC-162A 200 mw, no clipping

Figure A5-14



ARTICULATION INDEX COMPUTATION CHART
FOR

ANB-H-1A Harvard Design Earcushions 5-B, o, 200 mw, no clipping

Figure A5-15

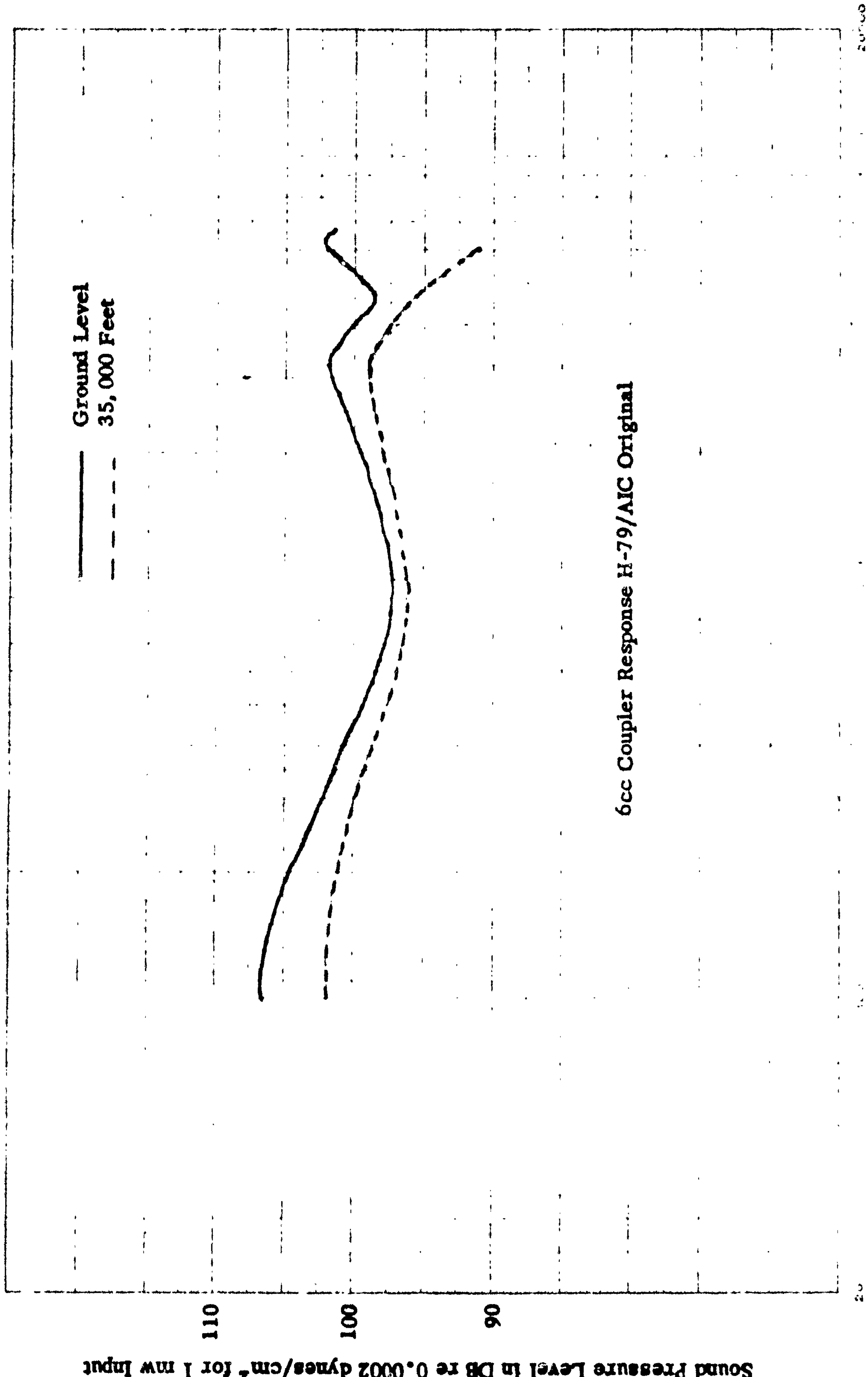
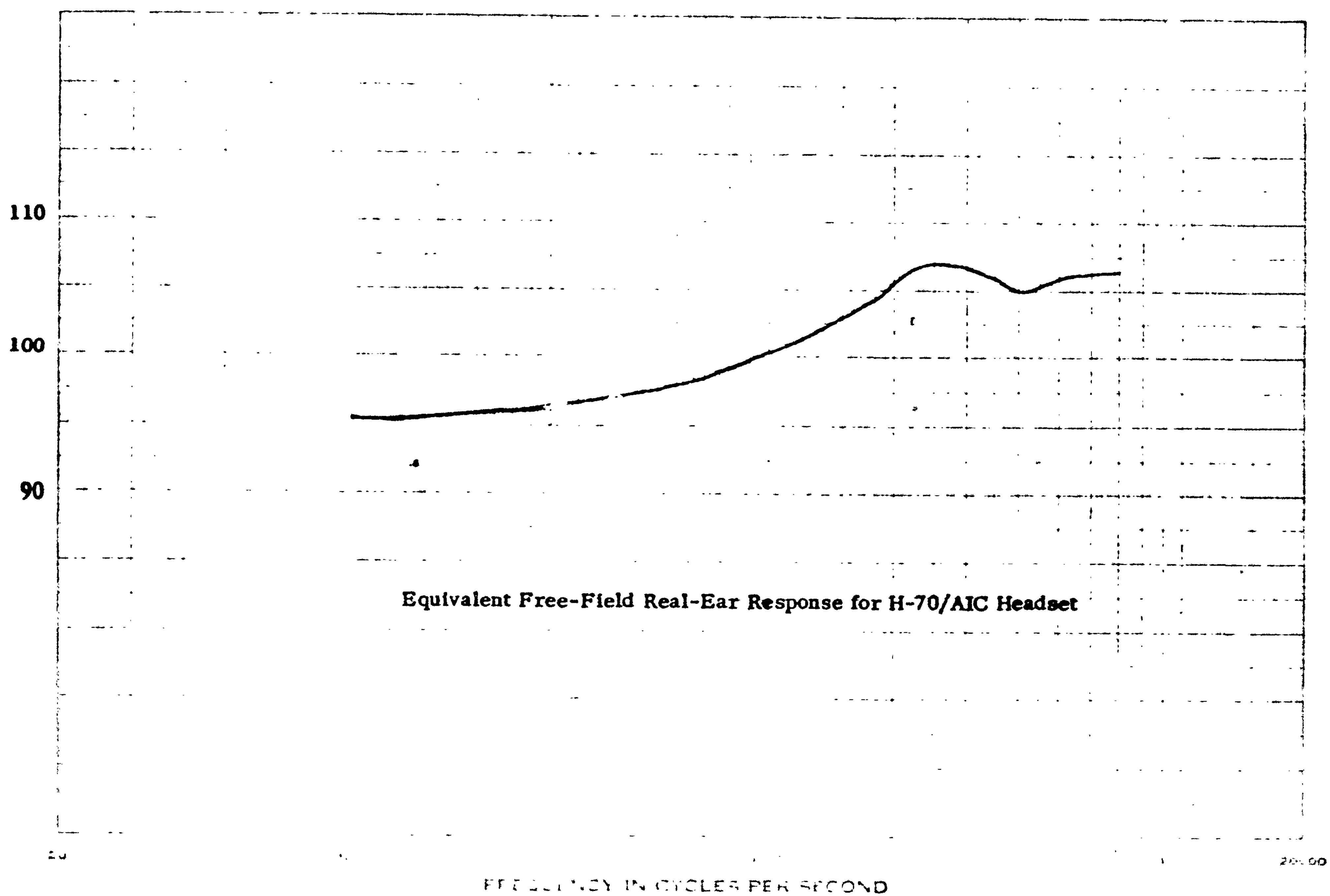

Ground Level
35,000 Feet
6cc Coupler Response H-79/AIC Original
FREQUENCY IN CYCLES PER SECOND
110
100
90
Sound Pressure Level in DB re 0.0002 dynes/cm^2 for 1 mw Input

Figure A5-16



Equivalent Free Field Sound Pressure Level in DB re 0.0002 dynes/cm^2
for 1 mw Input

Figure A5-17

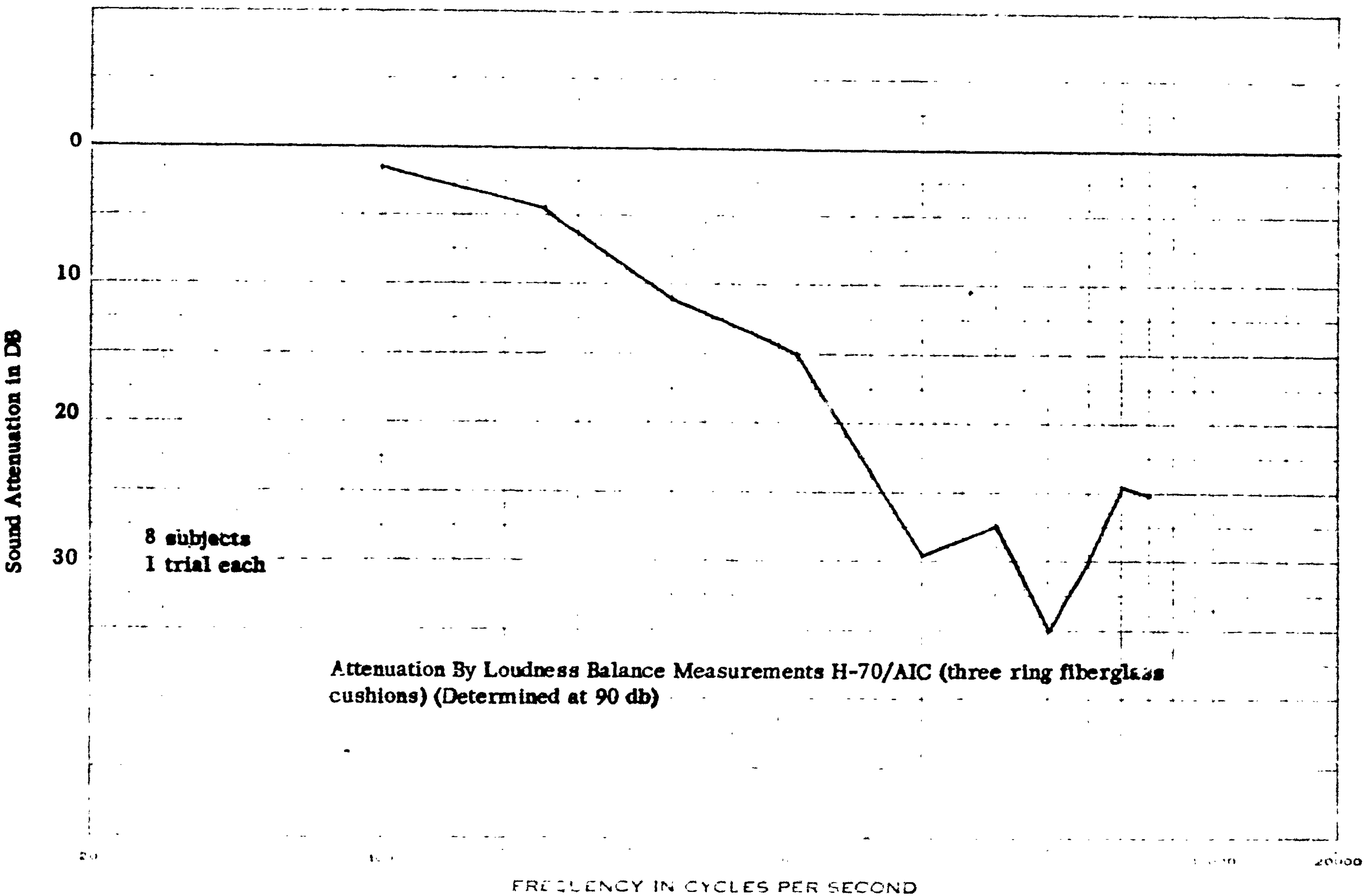
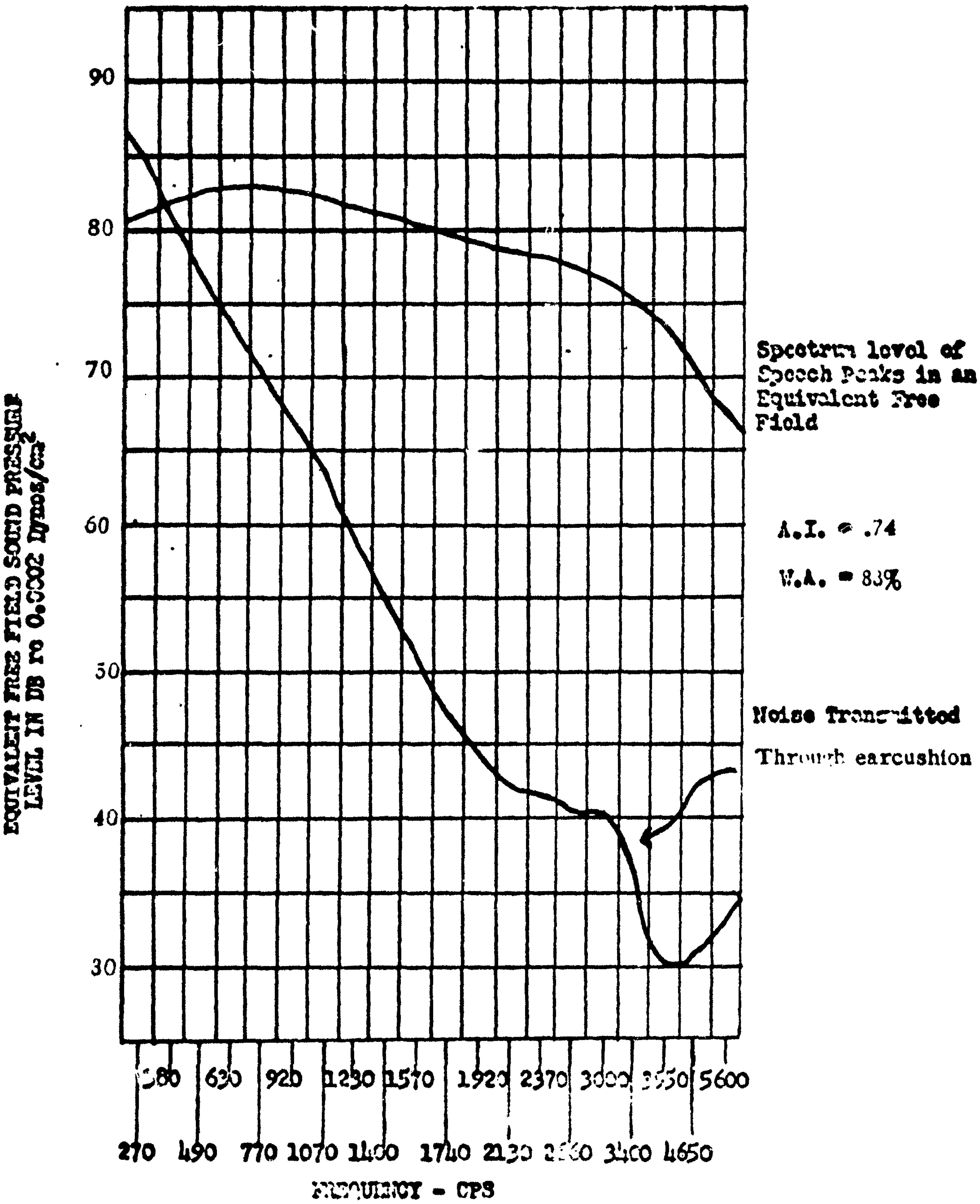
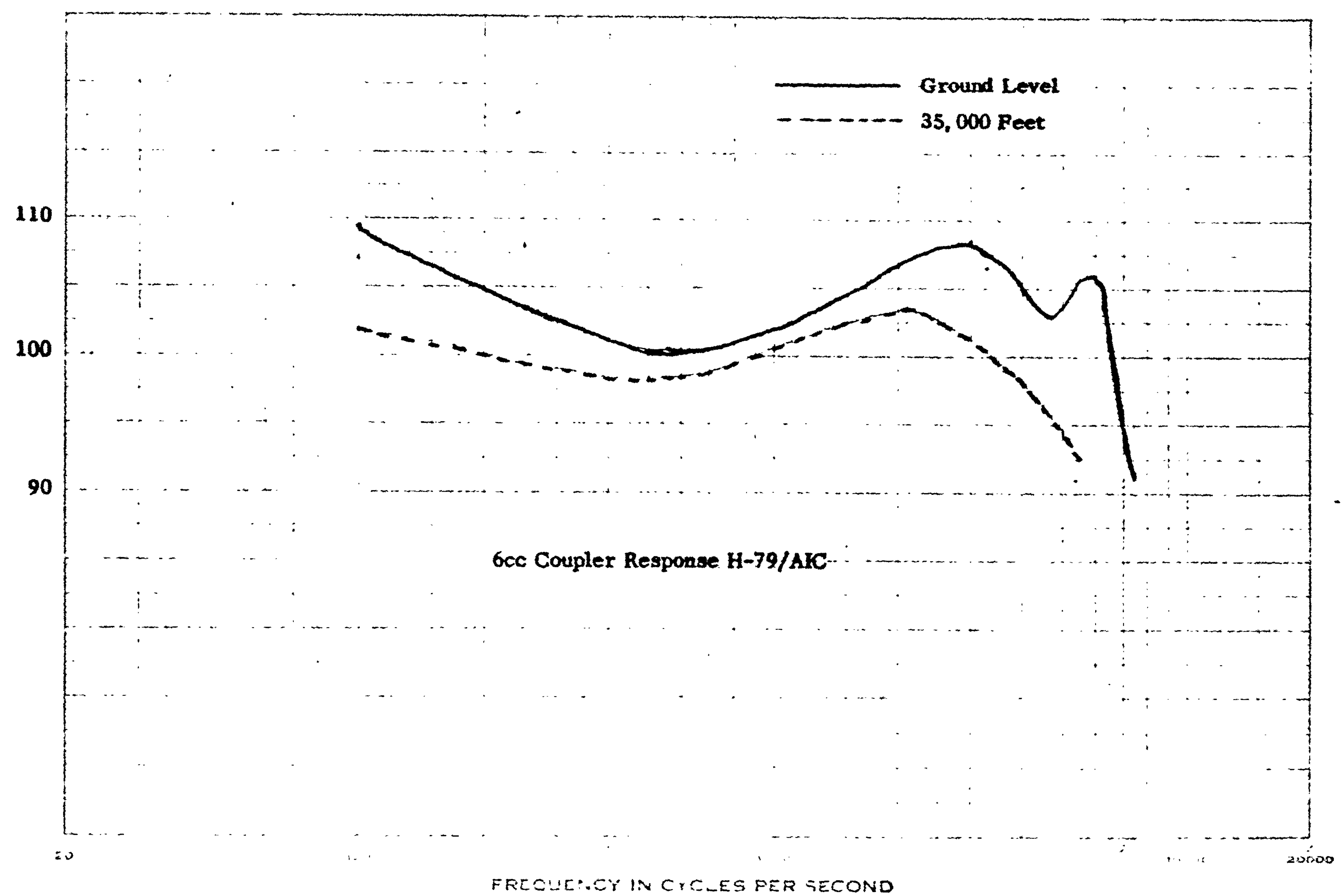

0
10
20
30
Sound Attenuation in DB
8 subjects
1 trial each
Attenuation By Loudness Balance Measurements H-70/AIC (three ring fiberglass cushions) (Determined at 90 db)
20000
FREQUENCY IN CYCLES PER SECOND

Figure A5-18



ARTICULATION INDEX COMPUTATION CHART
FOR
H-70/AIC Headset - Three Ring Fiberglass Earcushion
(200 mw, no clipping)

Figure A5-19



Sound Pressure Level in DB re 0.0002 dynes/cm^2 per 1 mw Input

Figure A5-20

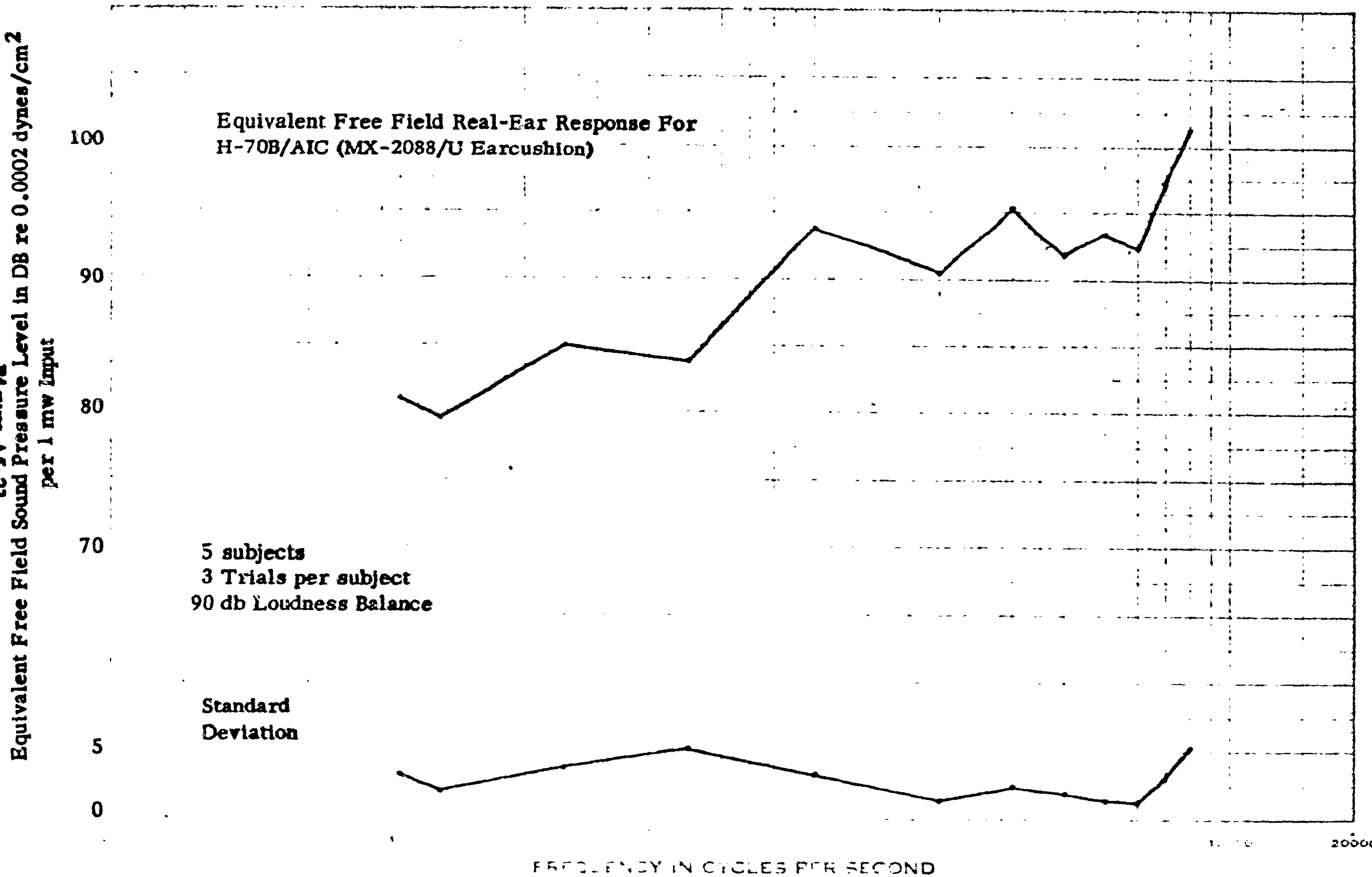

Equivalent Free Field Real-Ear Response For
H-70B/AIC (MX-2088/U Earcushion)
5 subjects
3 Trials per subject
90 db Loudness Balance
Standard
Deviation
100
90
80
70
5
0
20000
FREQUENCY IN CYCLES PER SECOND
Equivalent Free Field Sound Pressure Level in DB re 0.0002 dynes/cm^2
per 1 mw Input

Figure A5-21

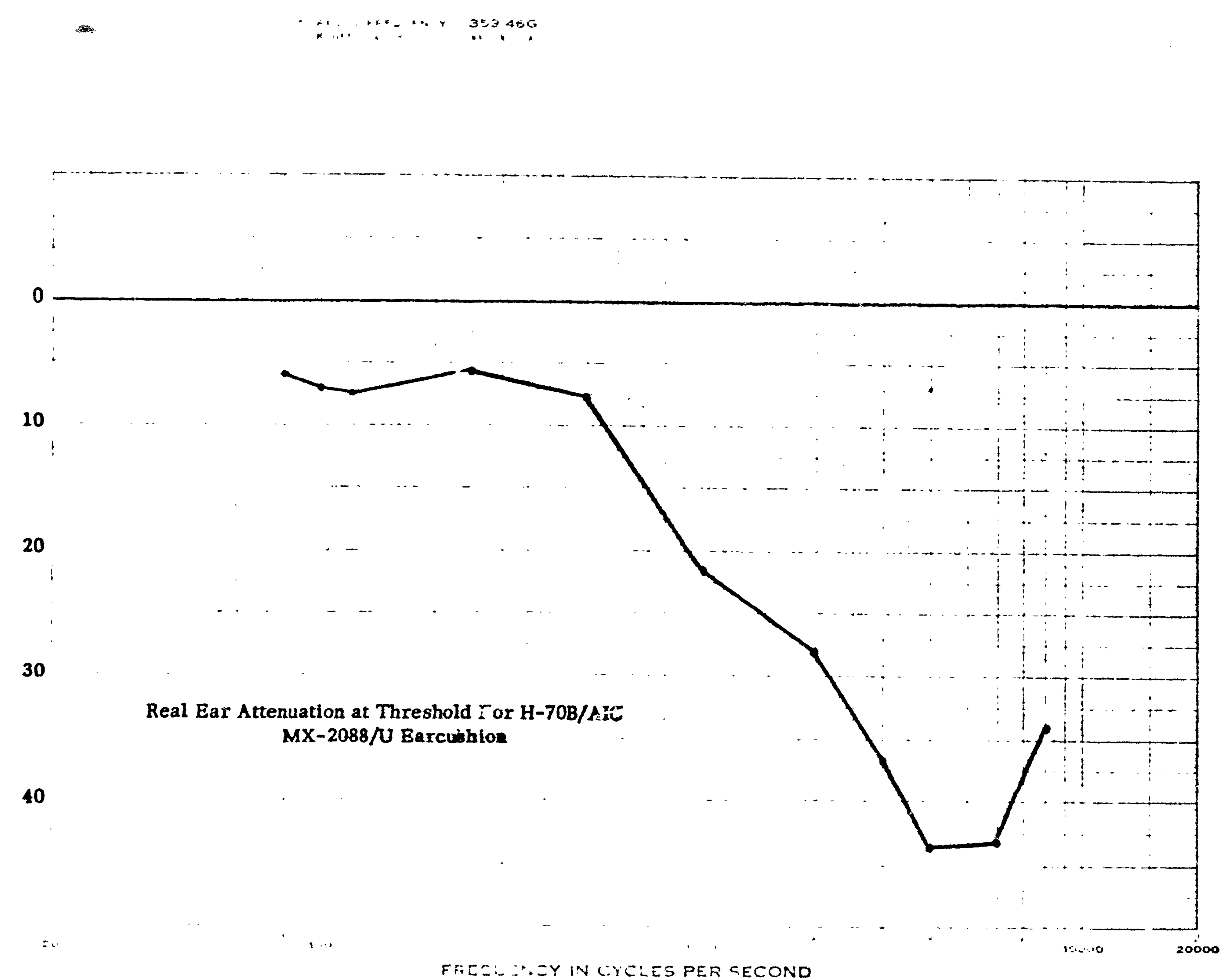
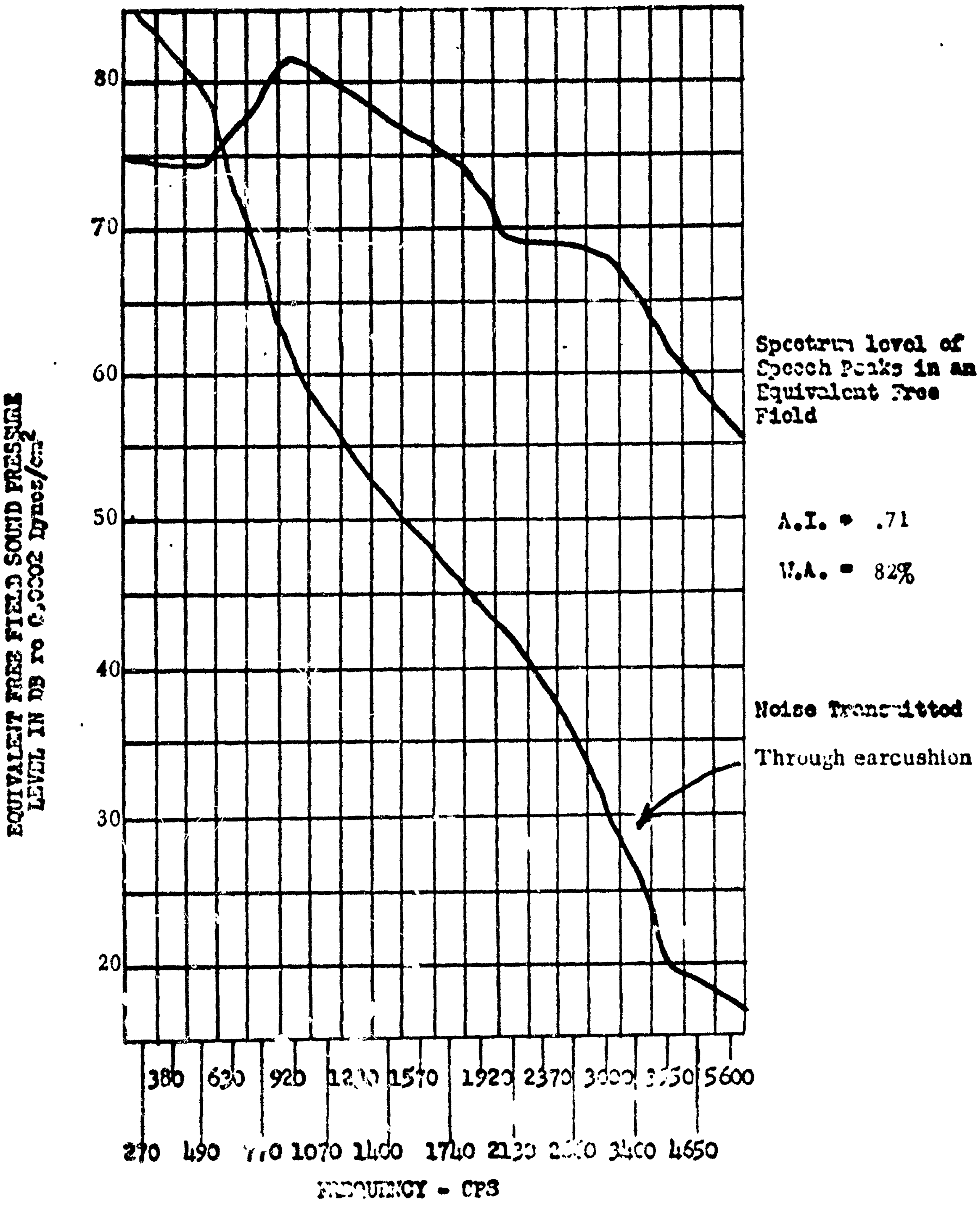


Figure A5-22
Sound Attenuation in DB



ARTICULATION INDEX COMPUTATION CHART
FOR

H-70B/AIC Headset MX-2088/U Earcushion, 200 mw, no clipping

Figure A5-23

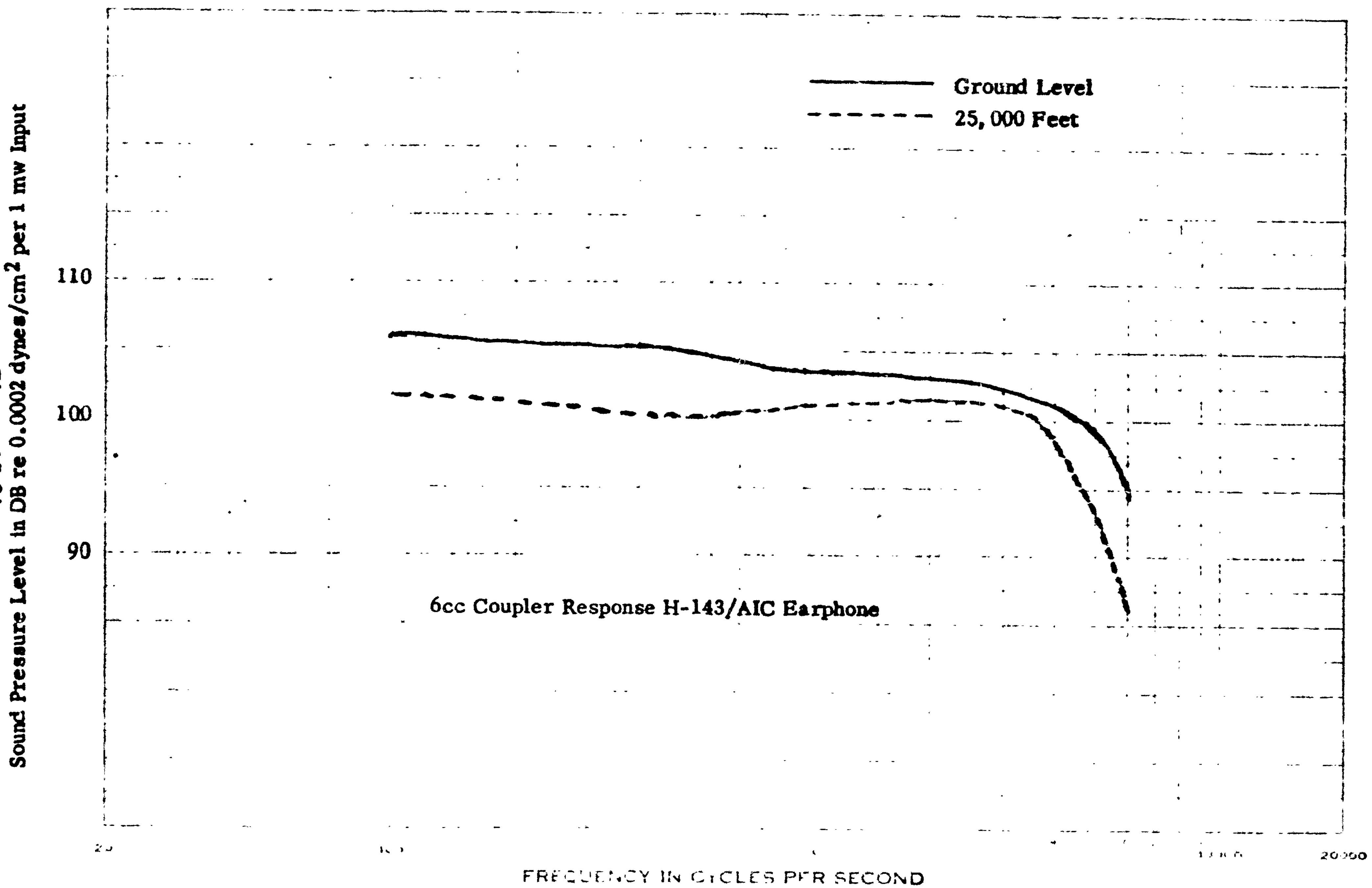

Ground Level
25, 000 Feet
110
100
90
6cc Coupler Response H-143/AIC Earphone
Sound Pressure Level in DB re 0.0002 dynes/cm² per 1 mw Input
20000
FREQUENCY IN CYCLES PER SECOND

Figure A5-24

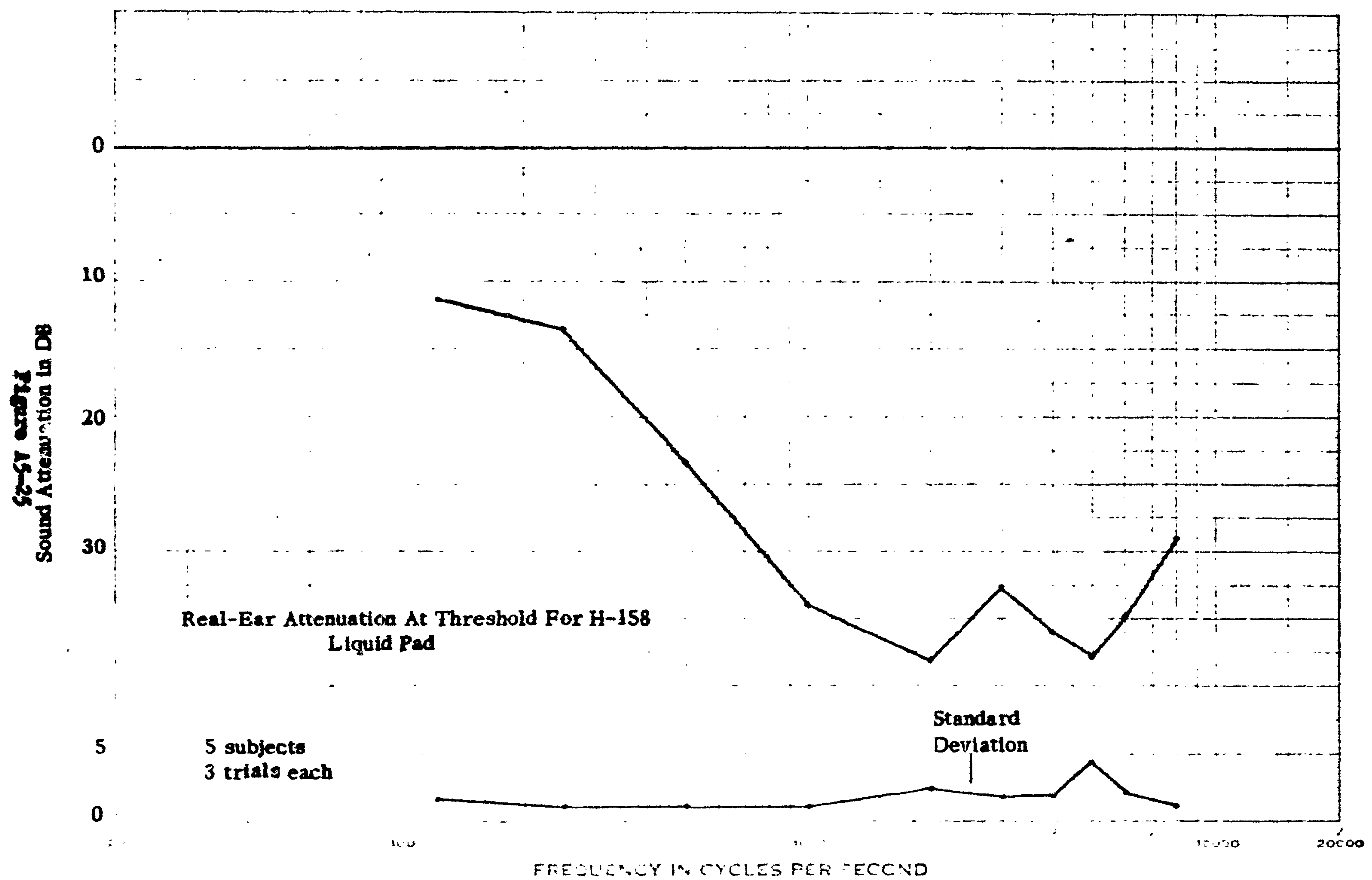

Real-Ear Attenuation At Threshold For H-158
Liquid Pad
5 subjects
3 trials each
Standard
Deviation
Sound Attenuation in DB
0
10
20
30
5
0
20000
FREQUENCY IN CYCLES PER SECOND

Figure 15-25

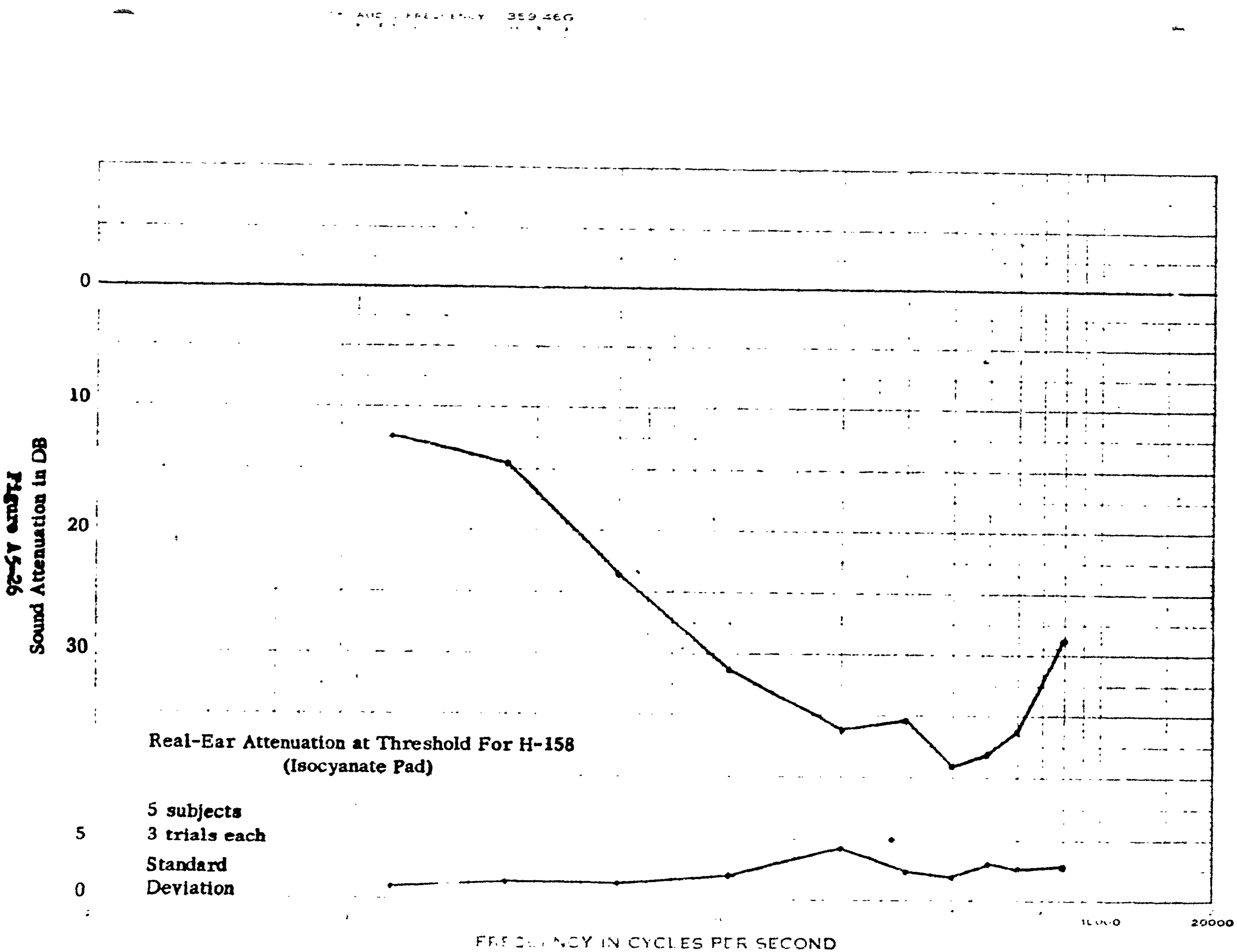

0
10
20
30
5
0
Sound Attenuation in DB
Real-Ear Attenuation at Threshold For H-158
(Isocyanate Pad)
5 subjects
3 trials each
Standard
Deviation
20000
FREQUENCY IN CYCLES PER SECOND

Figure A5-26

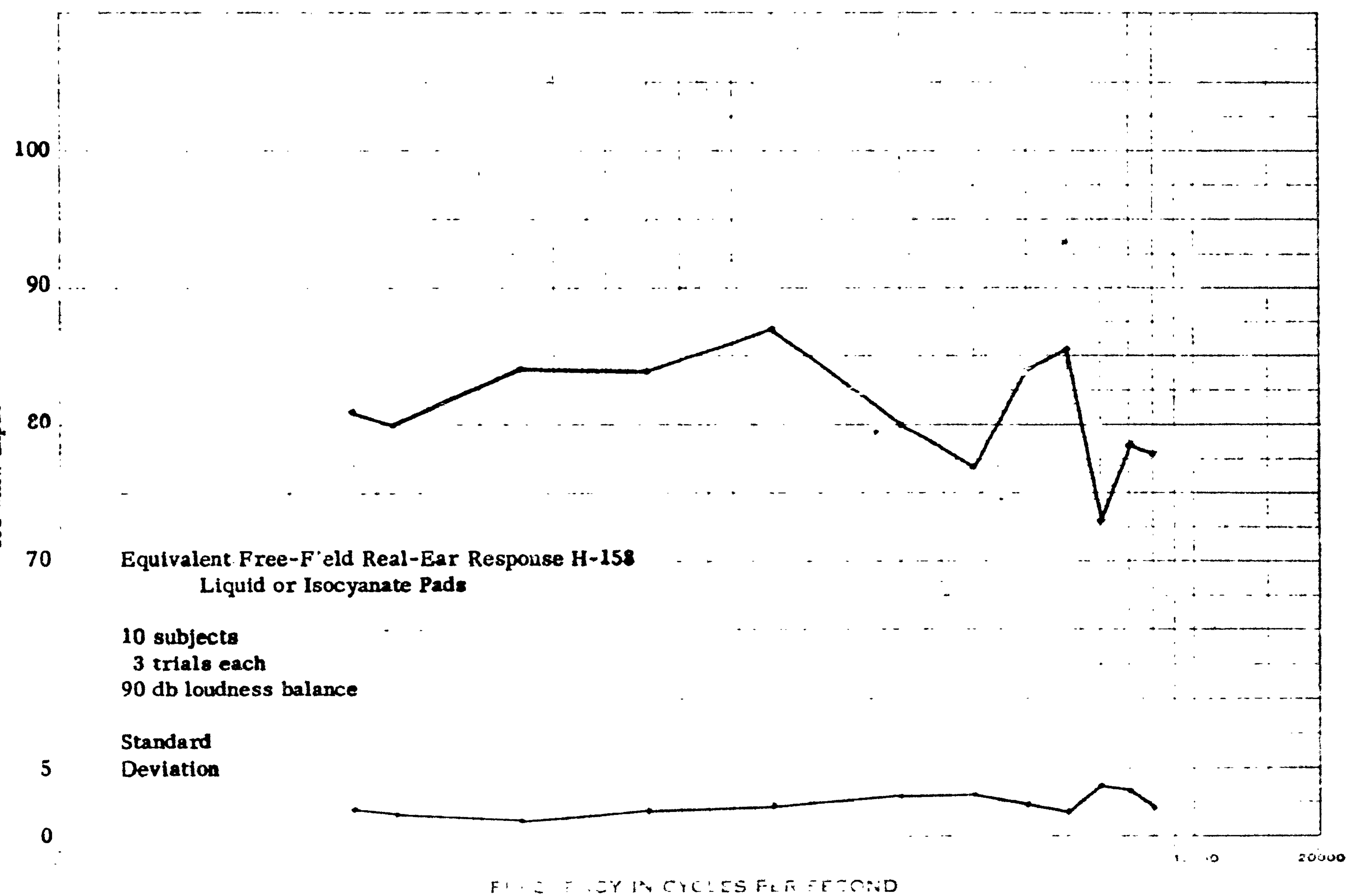
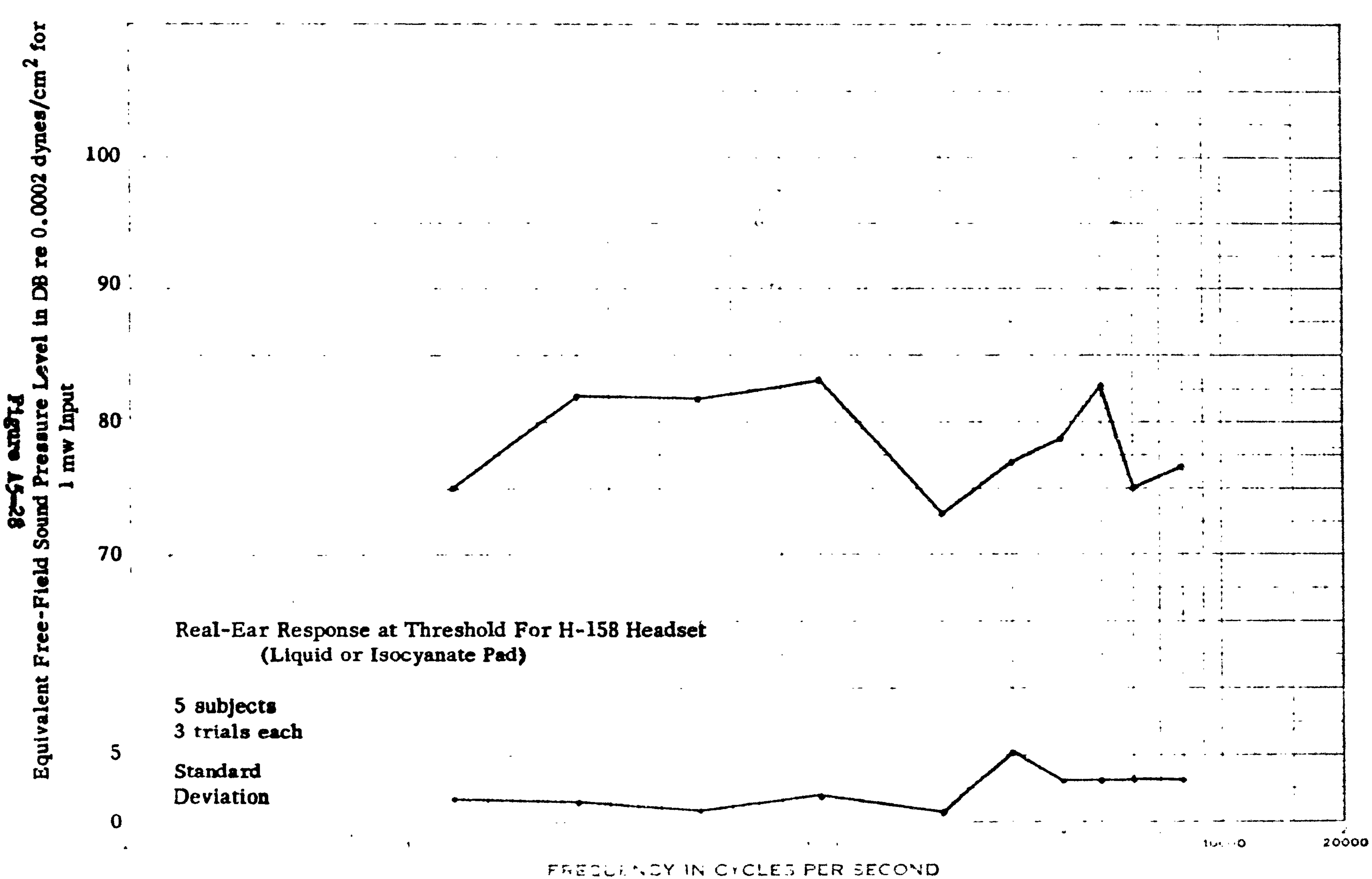

Equivalent Free-Field Sound Pressure Level in DB re 0.0002 dynes/cm2
for 1mw Input
100
90
80
70
5
0
Equivalent Free-F'eld Real-Ear Response H-158
Liquid or Isocyanate Pads
10 subjects
3 trials each
90 db loudness balance
Standard
Deviation
20000

Figure A5-27



Equivalent Free-Field Sound Pressure Level in DB re 0.0002 dynes/cm² for
1 mw Input
100
90
80
70
5
0
Real-Ear Response at Threshold For H-158 Headset
(Liquid or Isocyanate Pad)
5 subjects
3 trials each
Standard
Deviation
20000
FREQUENCY IN CYCLES PER SECOND

Figure A5-28

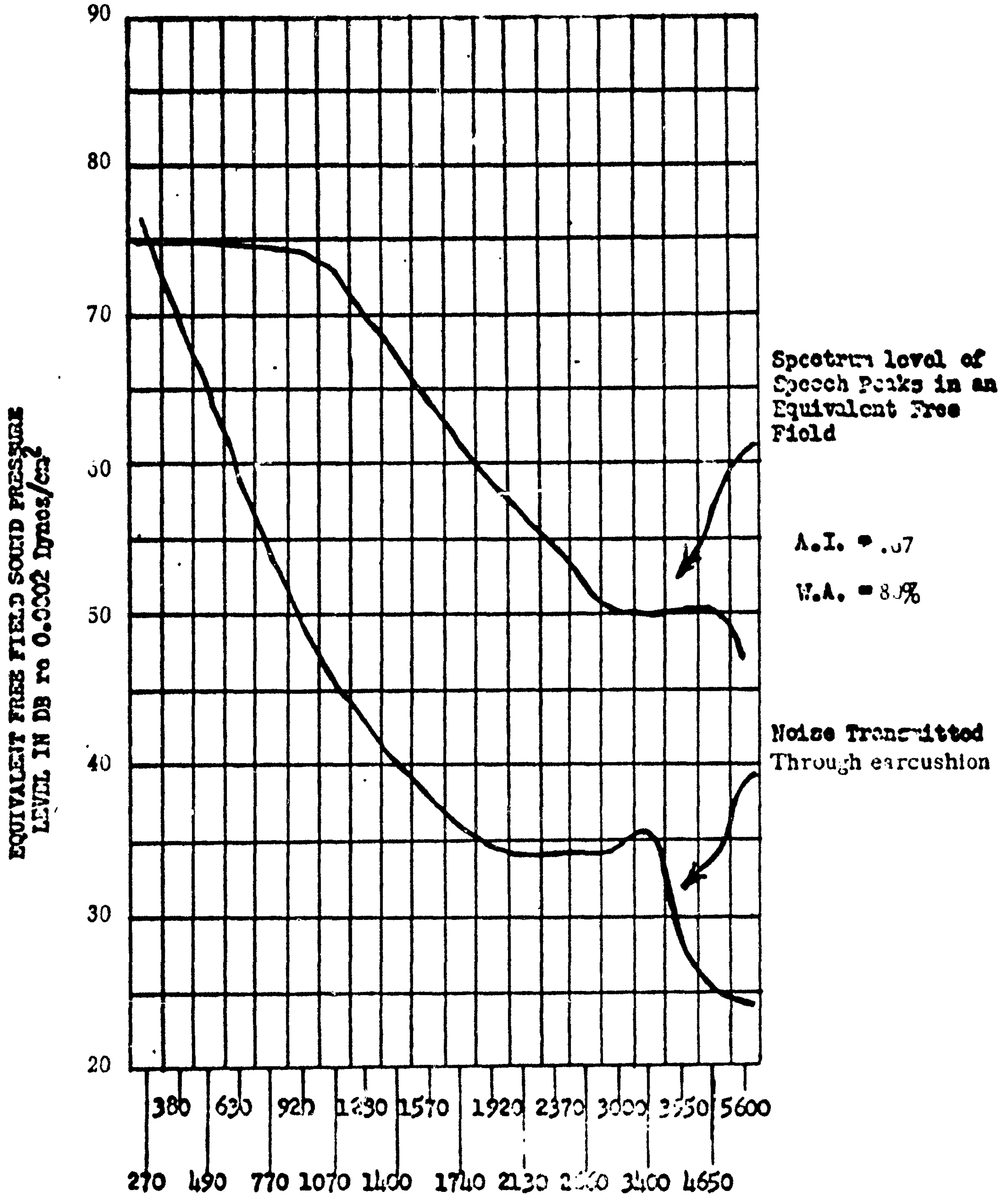
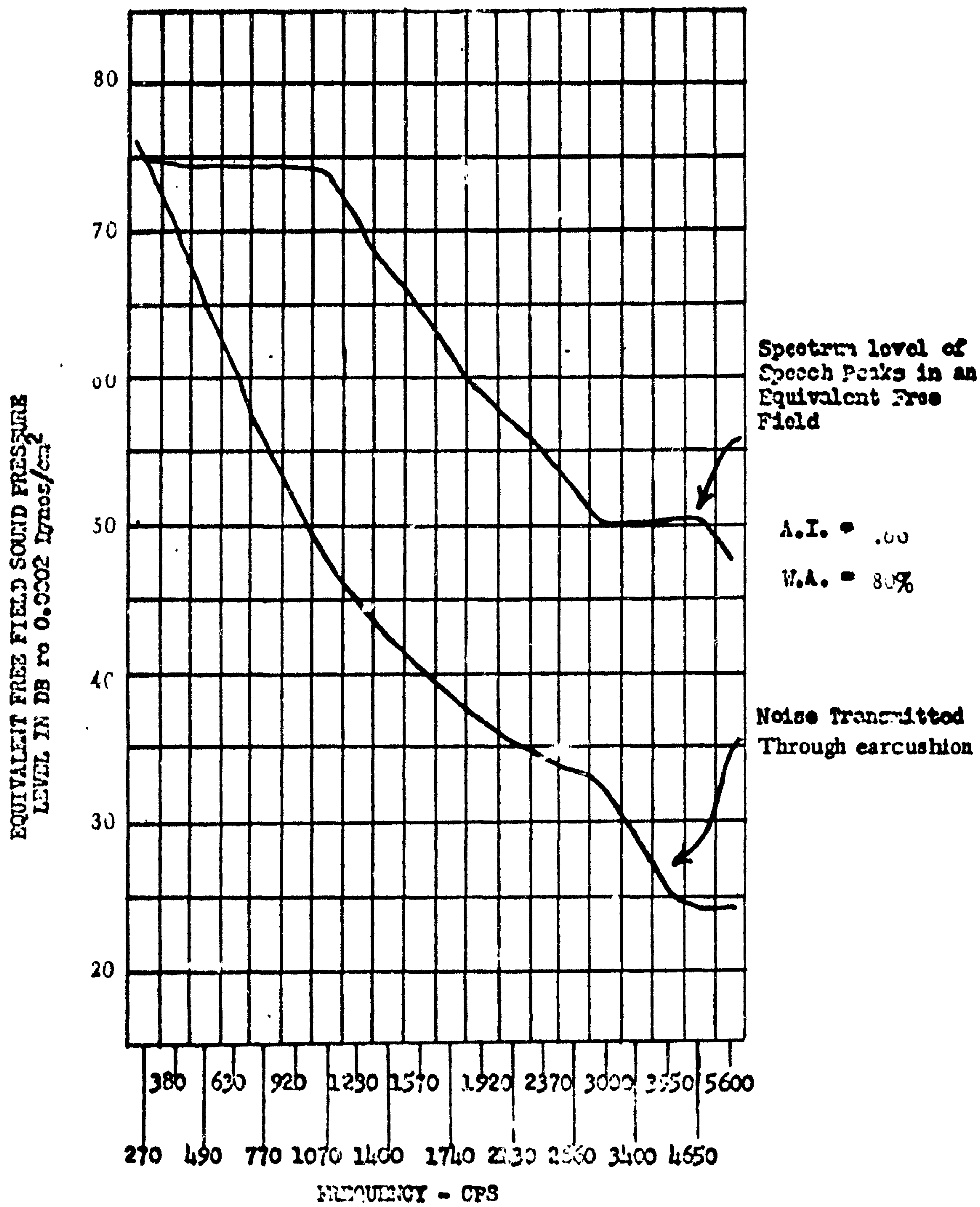


FIGURE A-29
ARTICULATION INDEX COMPUTATION CHART
FOR

H-158 Headset Liquid Pad, 200 mw, no clipping

Figure A5-29



ARTICULATION INDEX COMPUTATION CHART FOR

H-158 Headset - Isocyanate-Foam Pad
(200mw, no clipping)

Figure A5-30

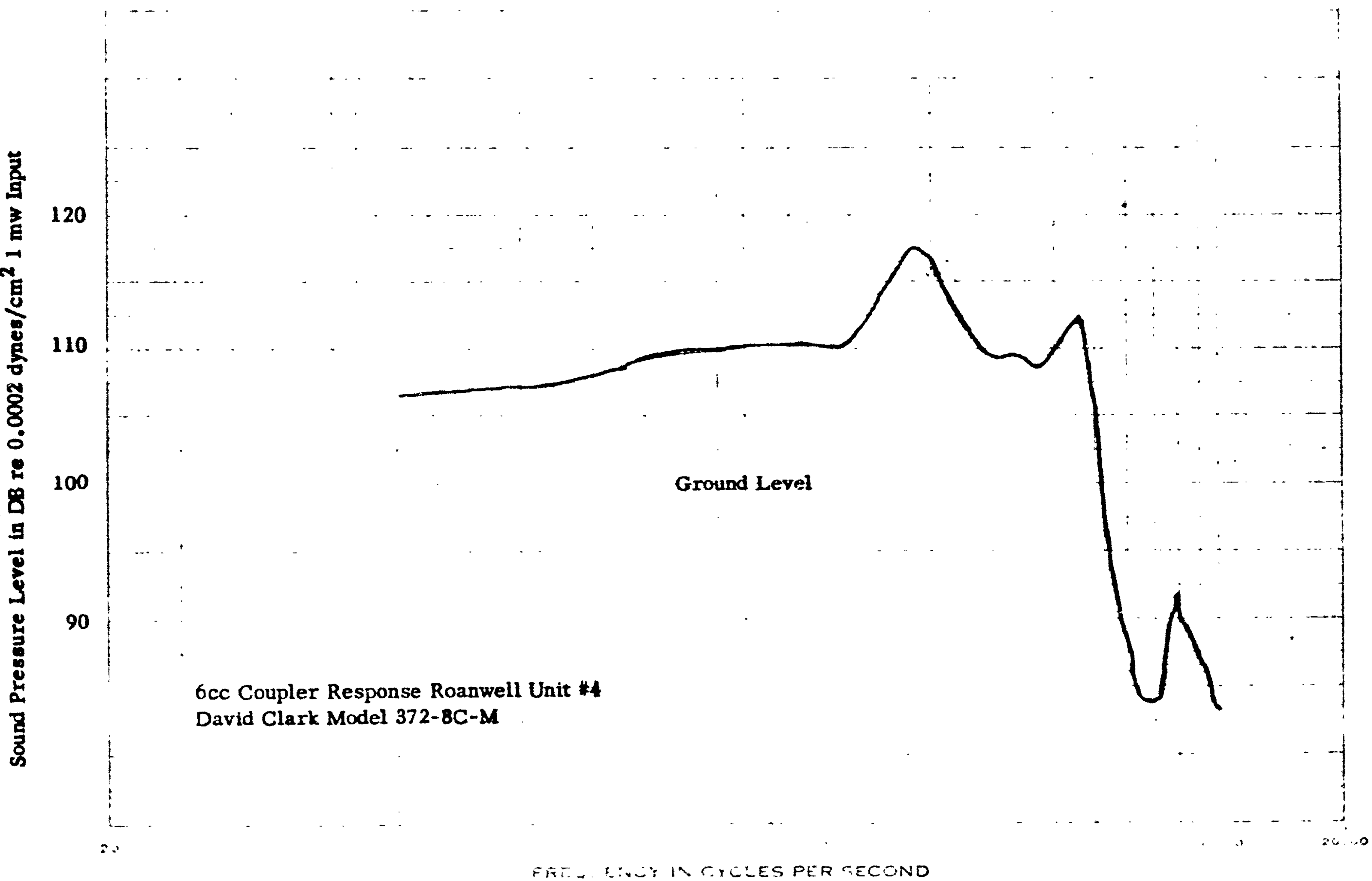

120
110
100
90
Sound Pressure Level in DB re 0.0002 dynes/cm2 1 mw Input
Ground Level
6cc Coupler Response Roanwell Unit #4
David Clark Model 372-8C-M
FREQUENCY IN CYCLES PER SECOND

Figure A5-31

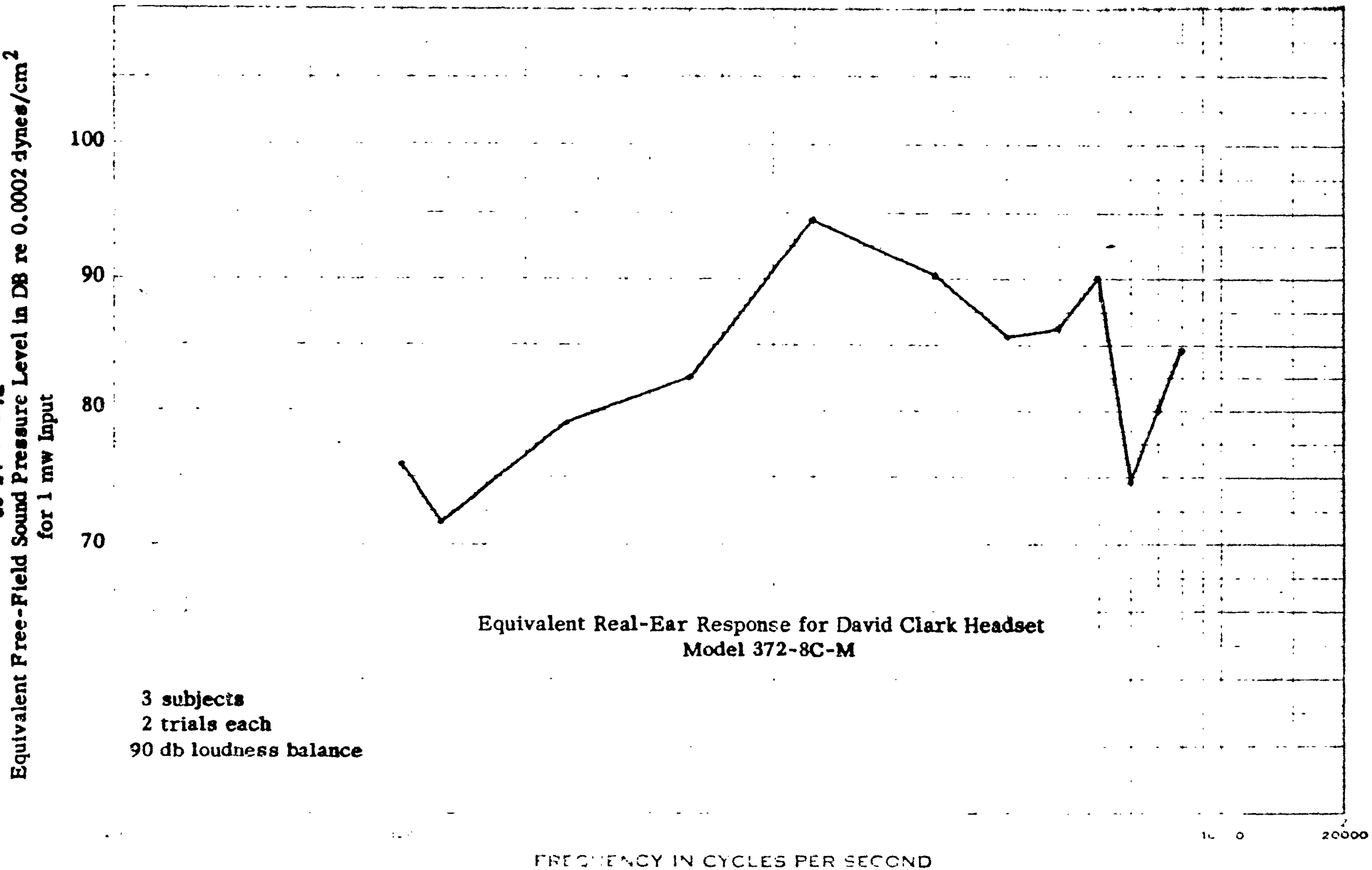
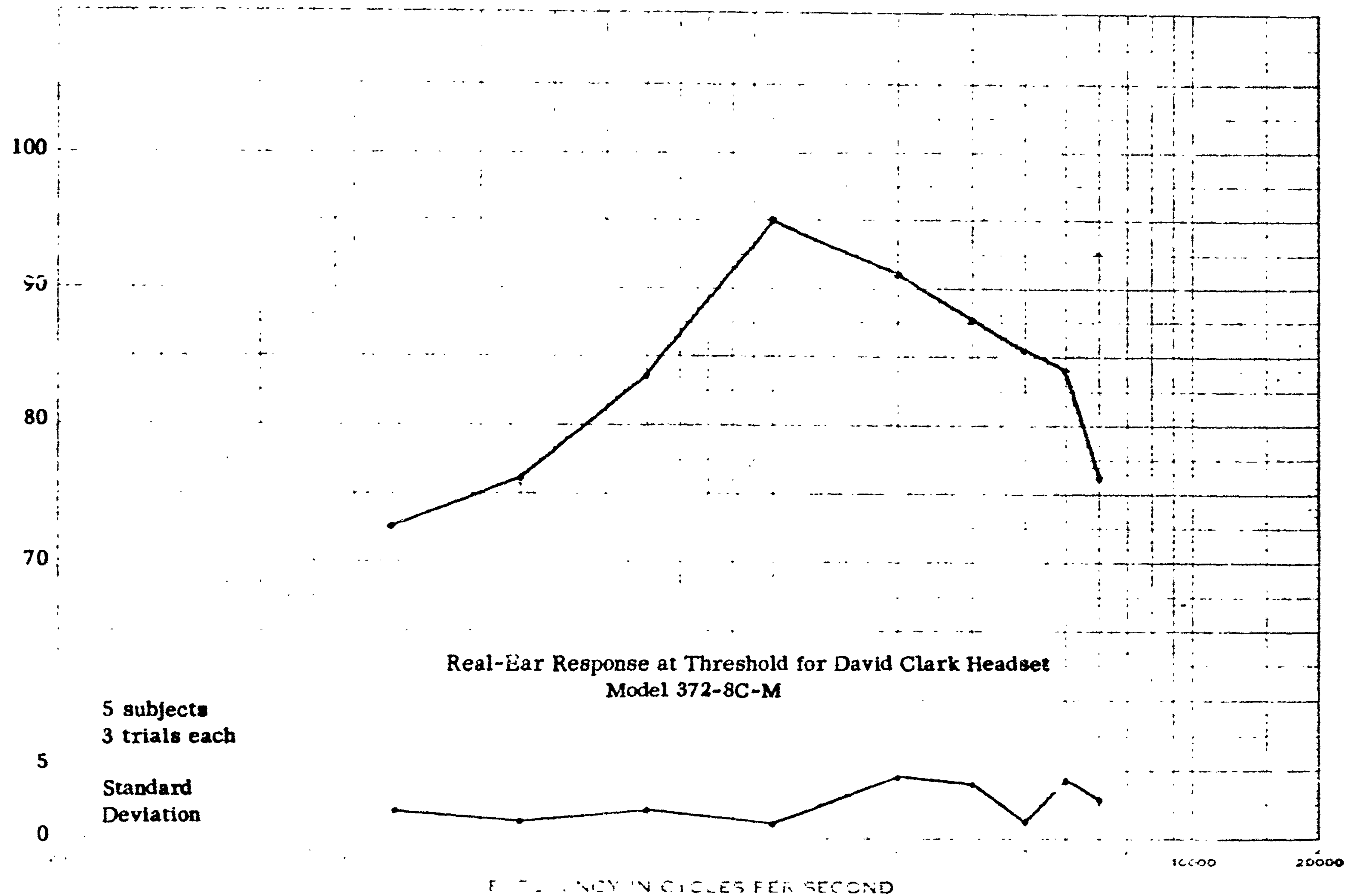

Equivalent Free-Field Sound Pressure Level in DB re 0.0002 dynes/cm^2 for 1 mw Input
100
90
80
70
Equivalent Real-Ear Response for David Clark Headset
Model 372-8C-M
3 subjects
2 trials each
90 db loudness balance
20000
FREQUENCY IN CYCLES PER SECOND

Figure A5-32



Equivalent Free-Field Sound Pressure Level in DB re 0.0002 dynes/cm^2 for 1 mw Input

Figure A5-33

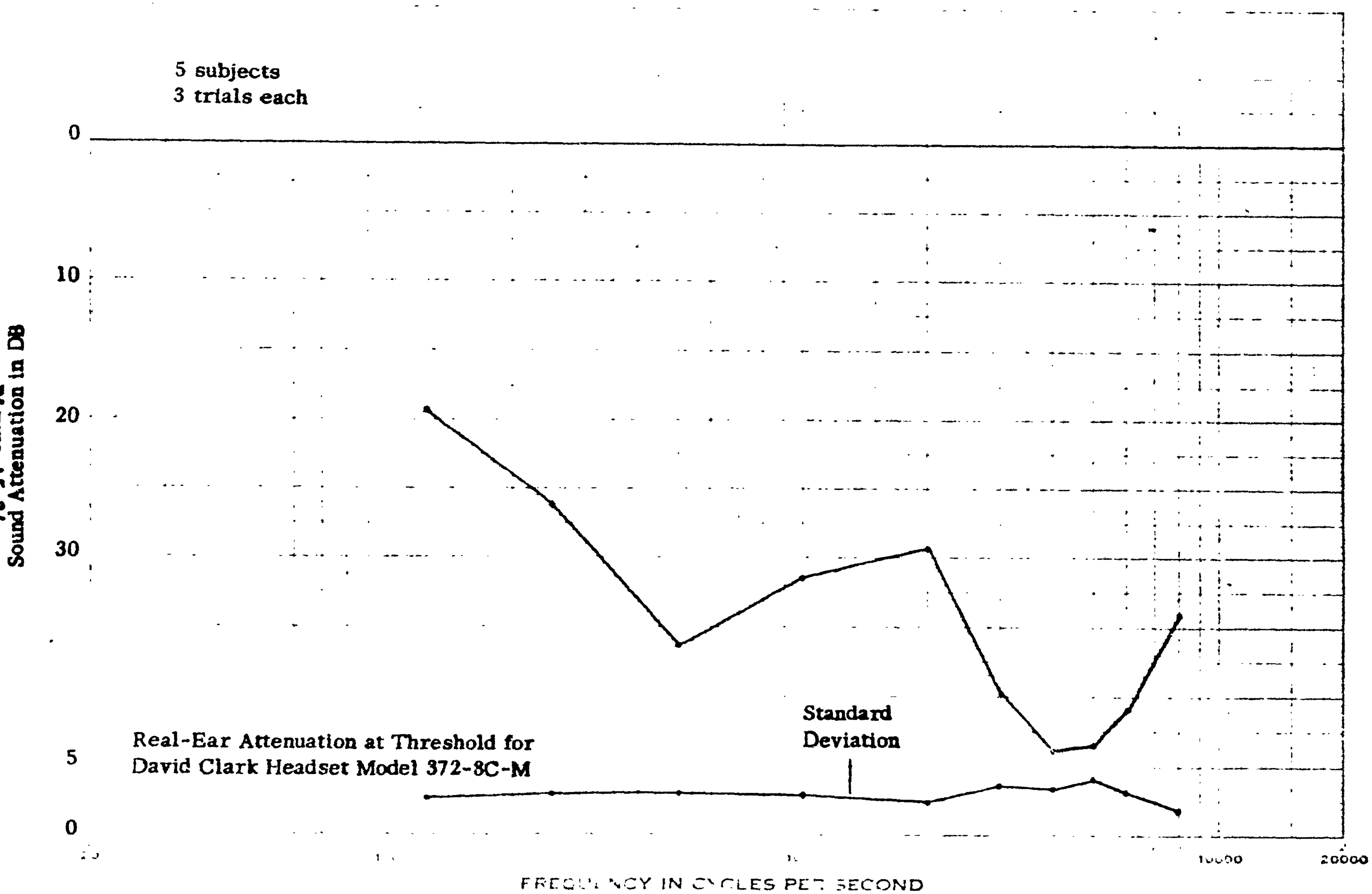
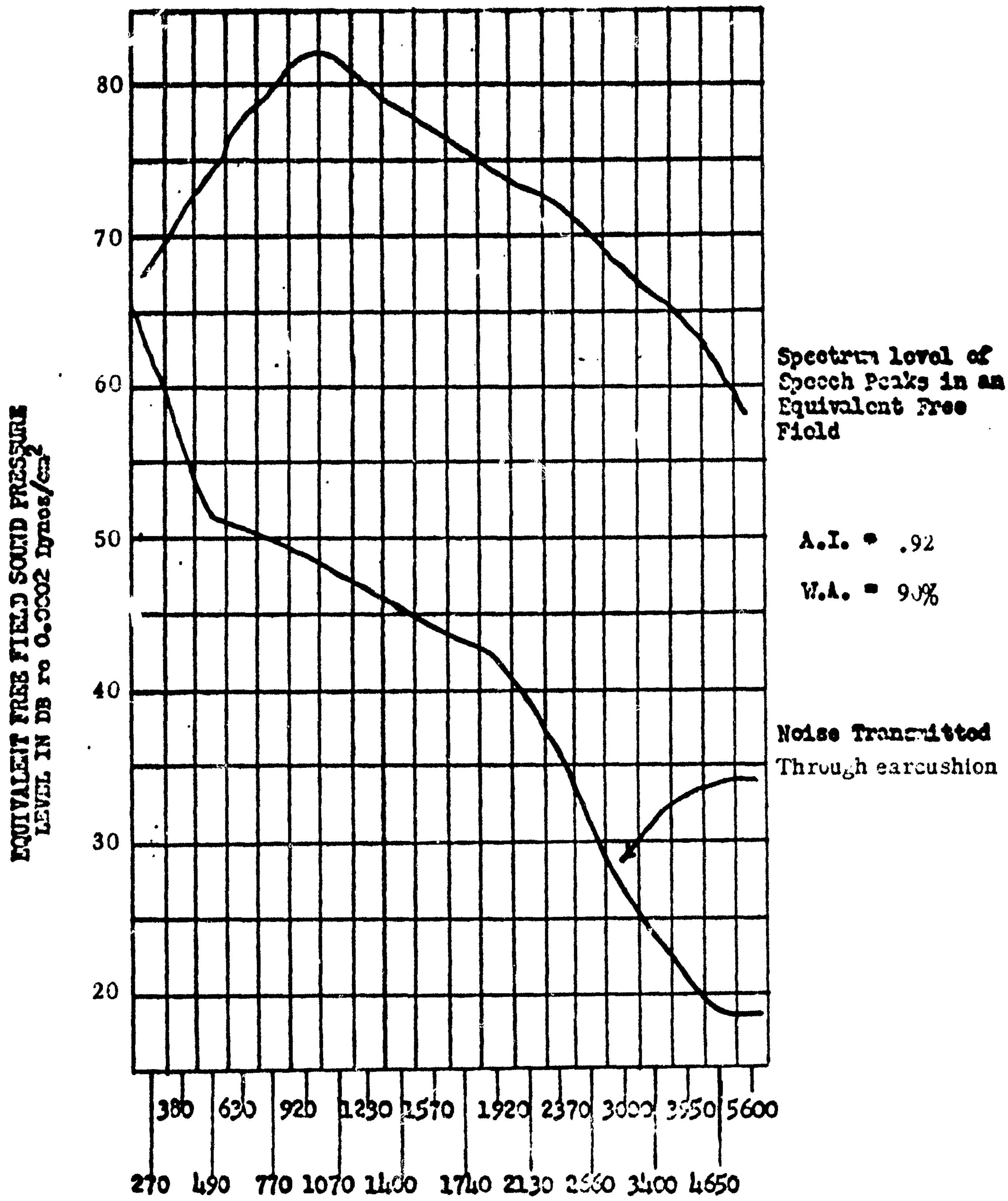

5 subjects
3 trials each
Sound Attenuation in DB
0
10
20
30
5
0
Real-Ear Attenuation at Threshold for
David Clark Headset Model 372-8C-M
Standard
Deviation
10000
20000

Figure A5-34



ARTICULATION INDEX COMPUTATION CHART FOR

David Clark Model 372-8C-M
(200 mw, no clipping)

Figure A5-35

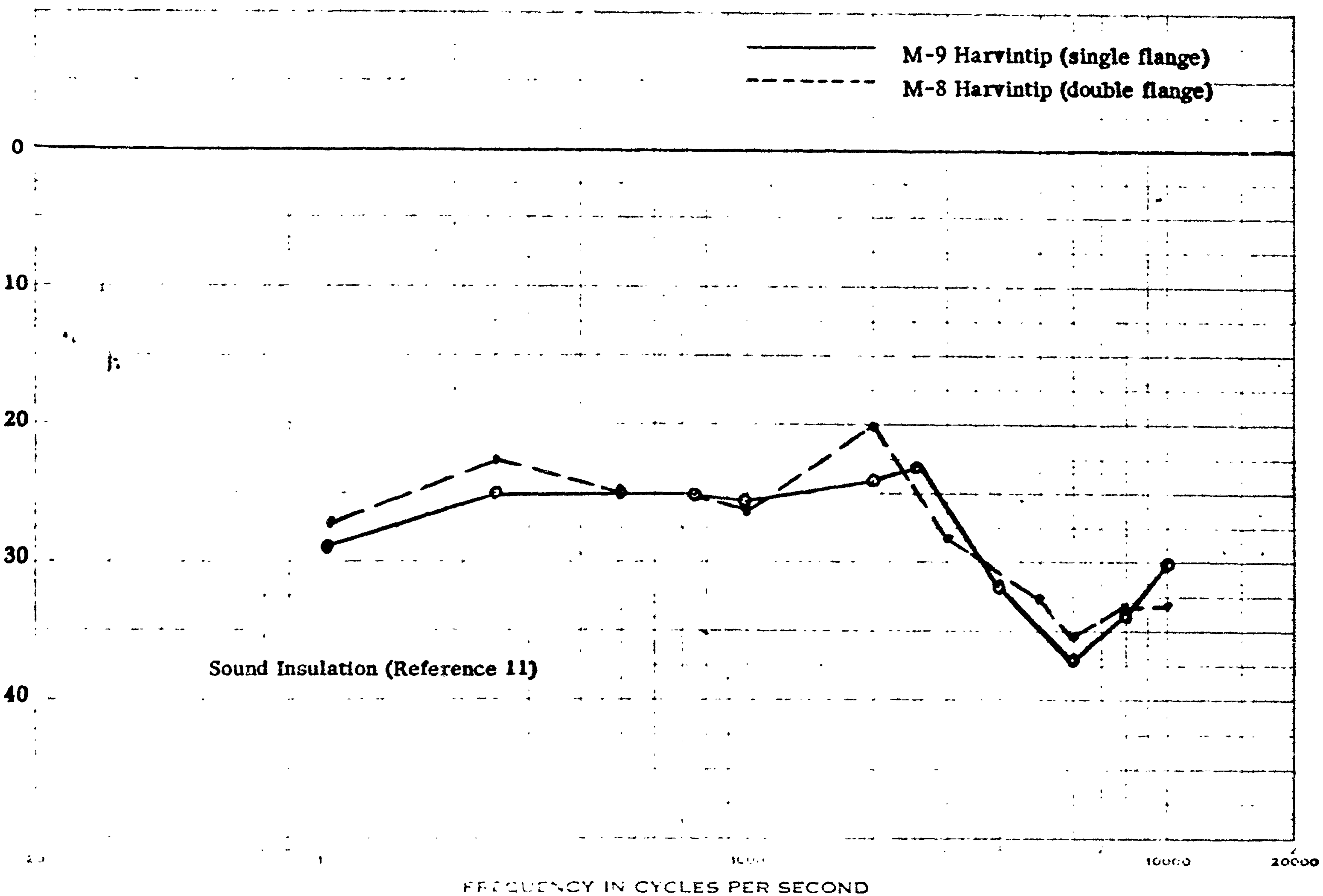

M-9 Harvintip (single flange)
M-8 Harvintip (double flange)
Sound Insulation (Reference 11)
0
10
20
30
40
Acoustic Insulation in DB
10000
20000
FREQUENCY IN CYCLES PER SECOND

Figure A5-36

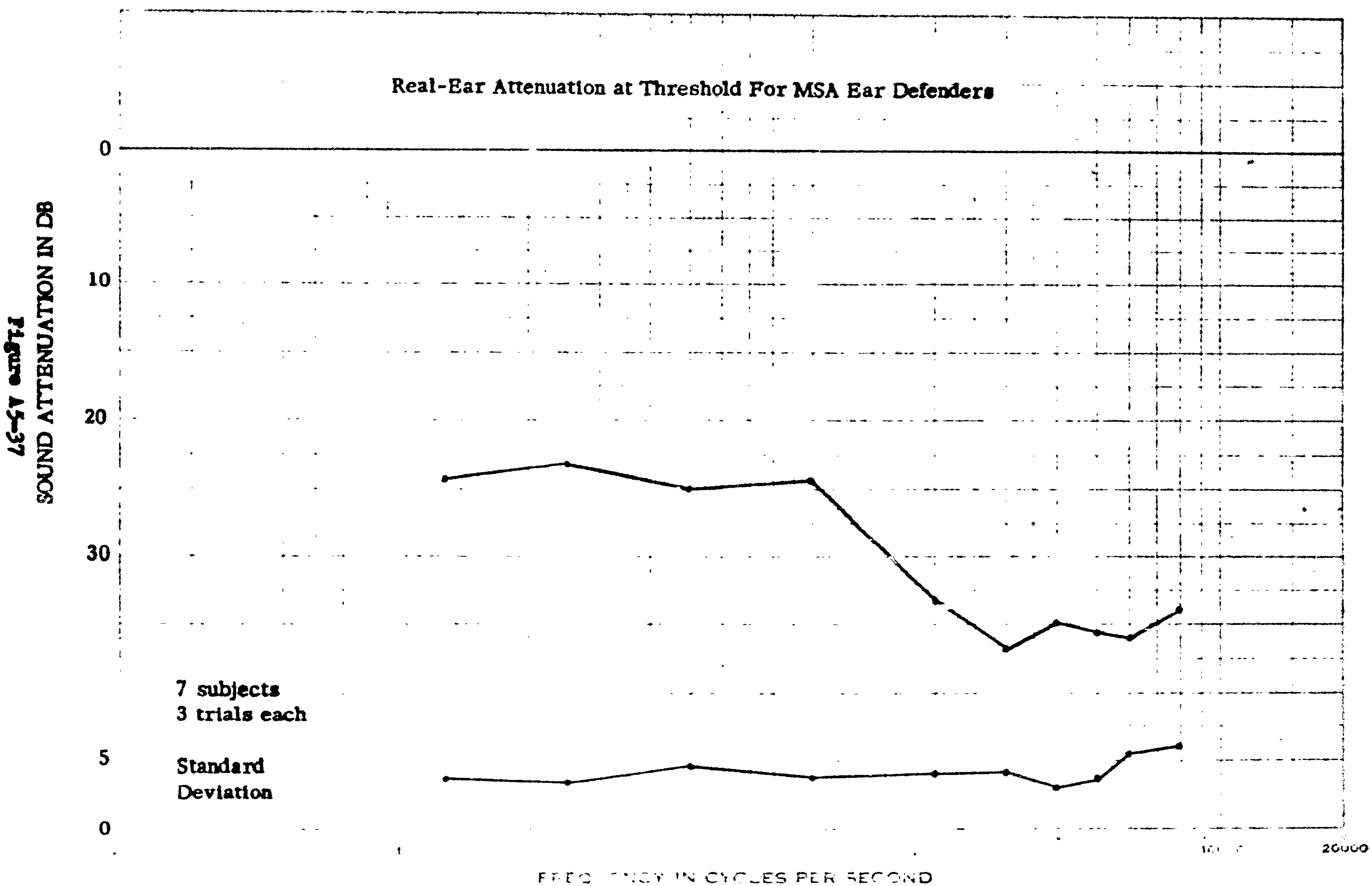
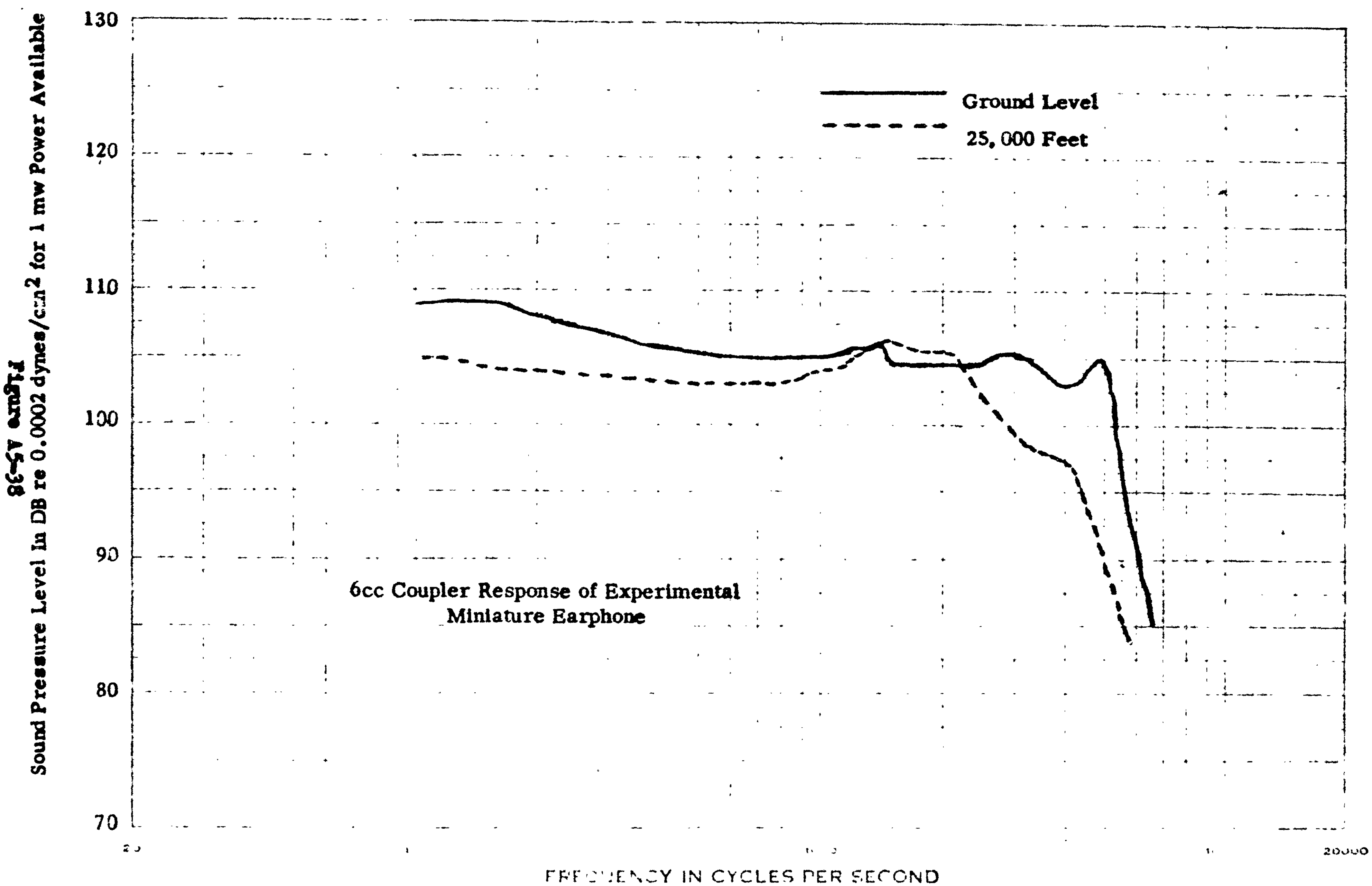
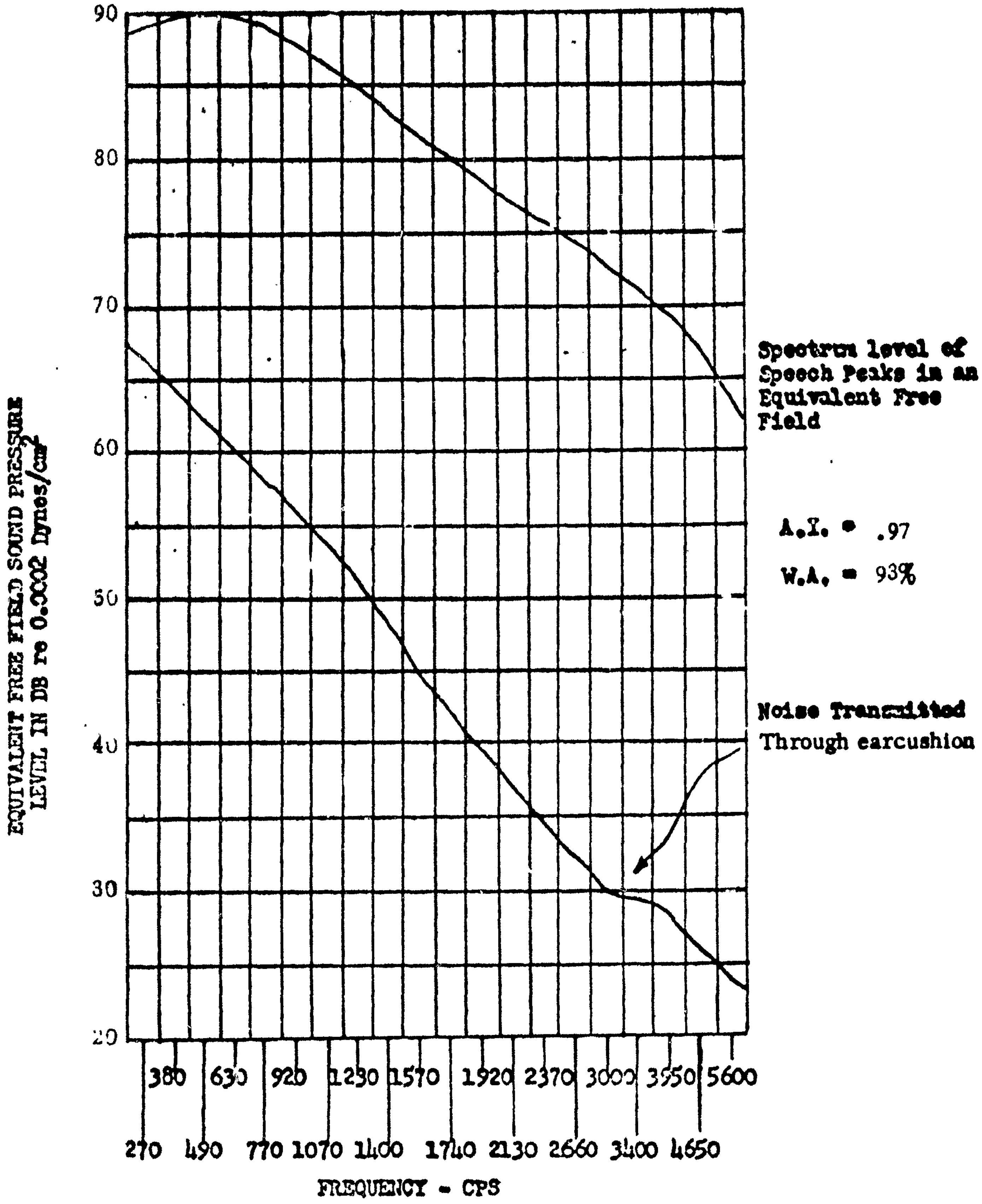

Real-Ear Attenuation at Threshold For MSA Ear Defenders
SOUND ATTENUATION IN DB
0
10
20
30
5
0
7 subjects
3 trials each
Standard
Deviation
20000
FREQUENCY IN CYCLES PER SECOND

Figure A5-37



130
120
110
100
90
80
70
Sound Pressure Level in DB re 0.0002 dynes/cm2 for 1 mw Power Available
Ground Level
25,000 Feet
6cc Coupler Response of Experimental
Miniature Earphone
20000
FREQUENCY IN CYCLES PER SECOND

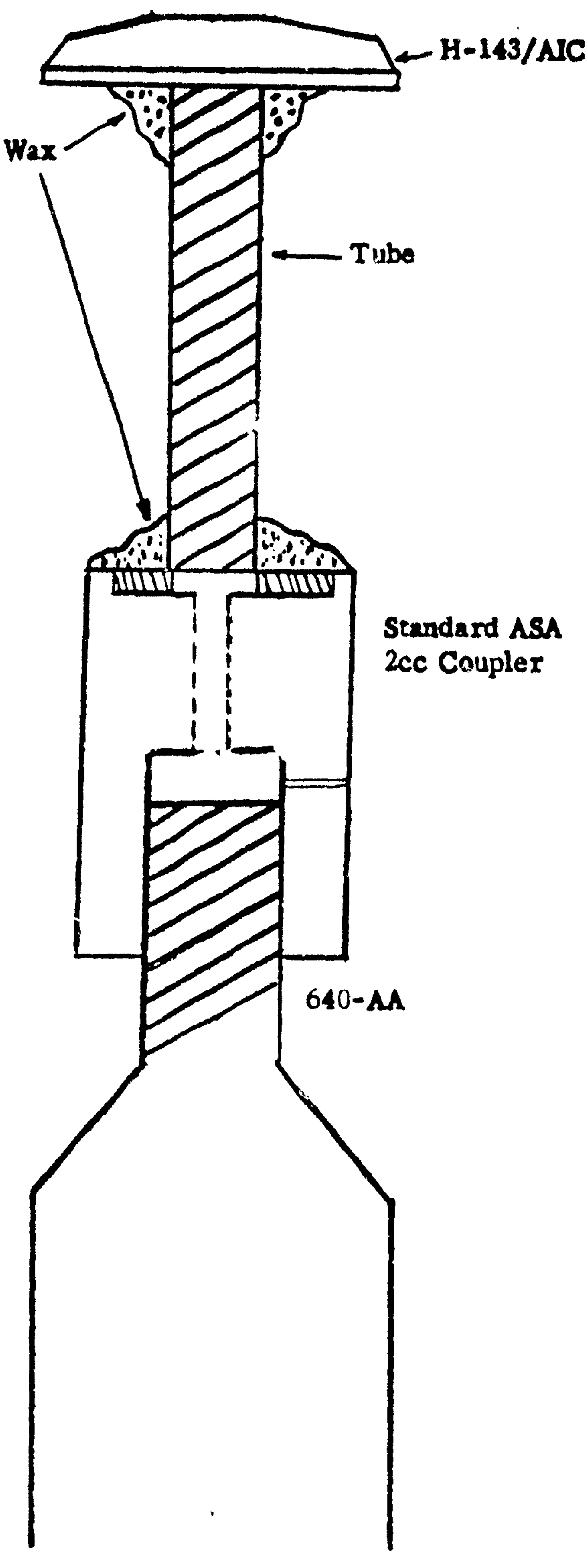
Figure A5-38



ARTICULATION INDEX COMPUTATION CHART FOR

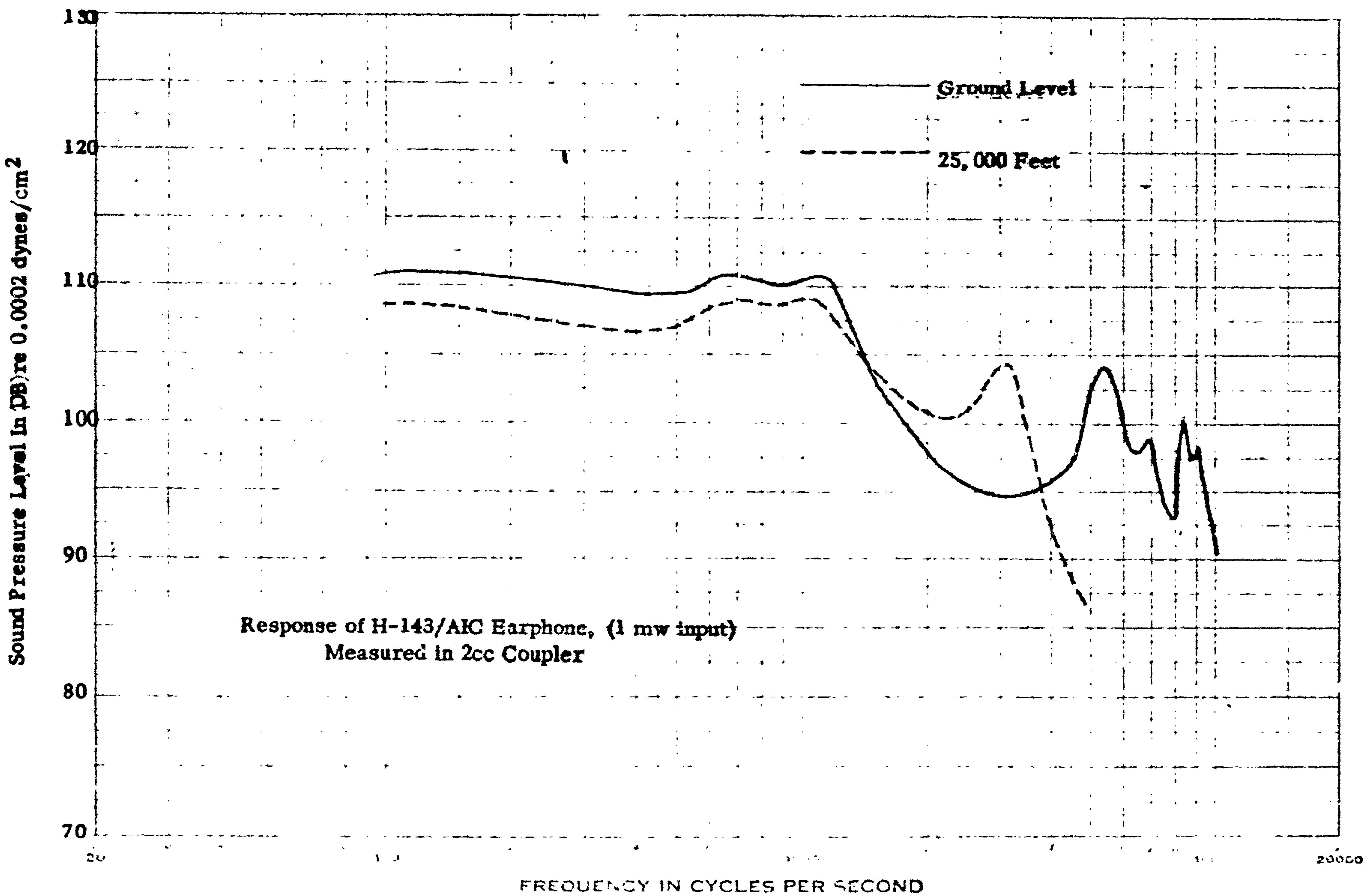
Miniature Earphone Coupled To MSA Ear Defender
200 mw, no clipping

Figure A5-39



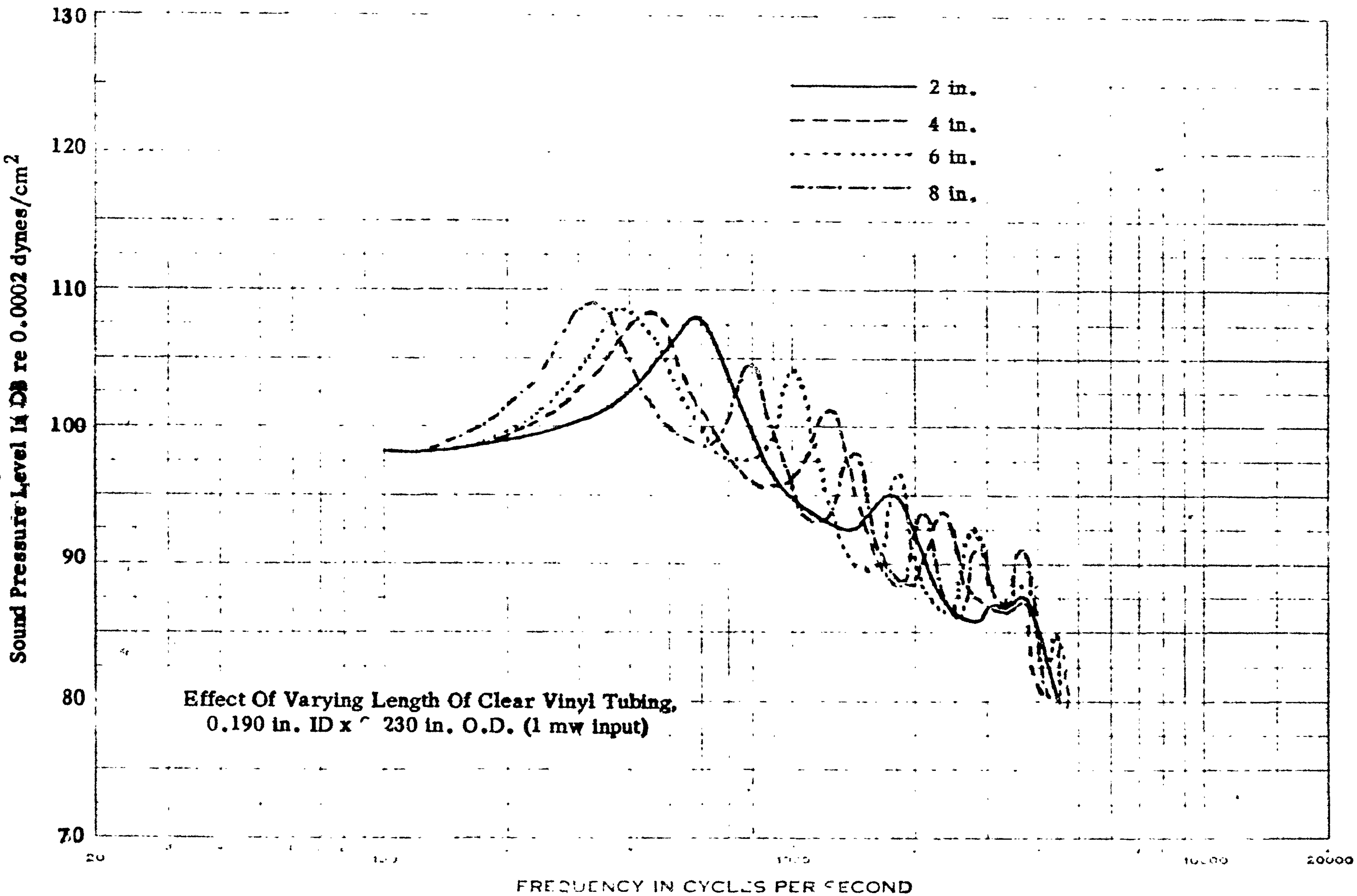
Set-Up For Measurement Of
Frequency Response Of Tubes

Figure A5-40



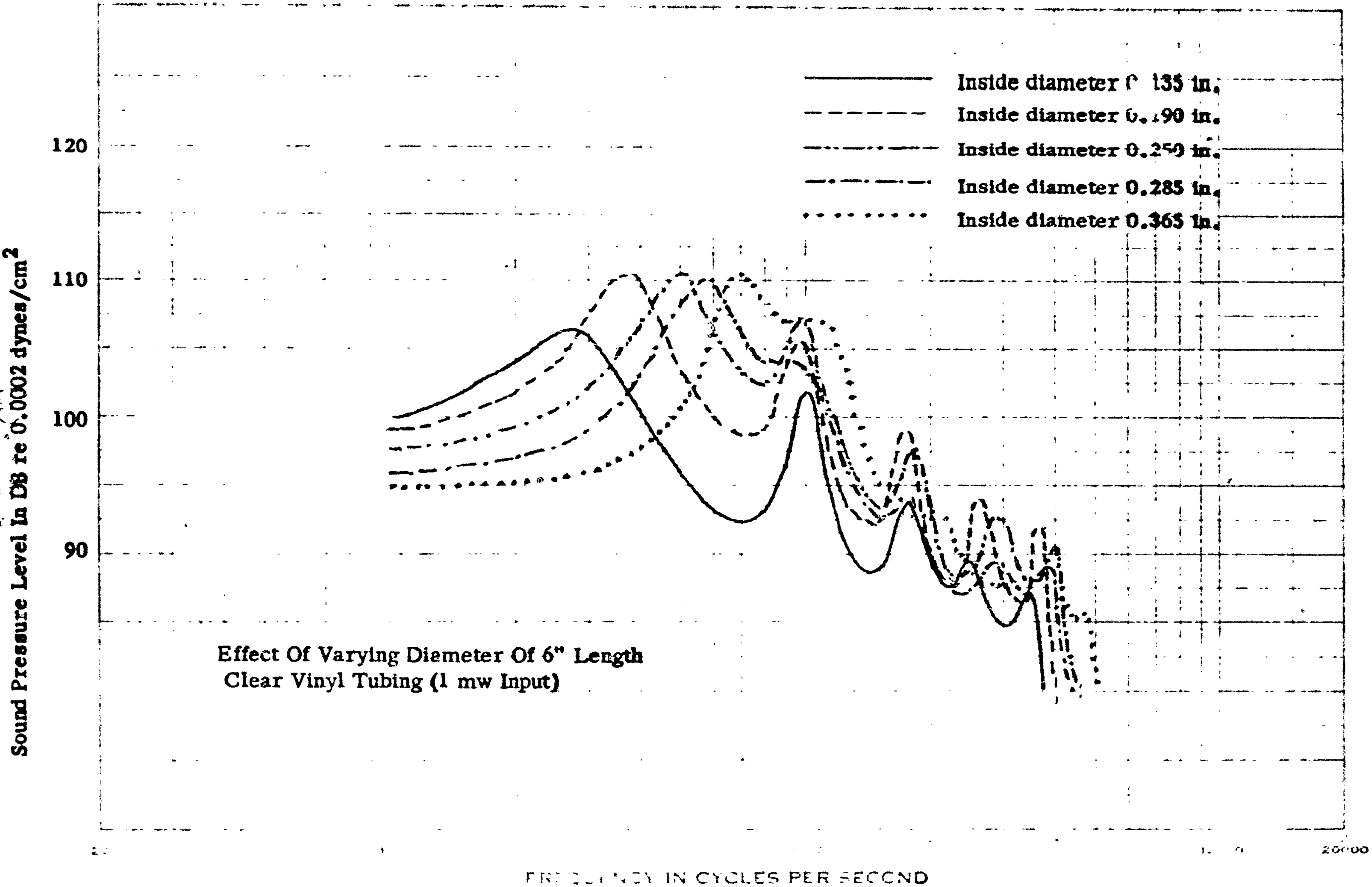
Ground Level
25,000 Feet
Response of H-143/AIC Earphone, (1 mw input)
Measured in 2cc Coupler
130
120
110
100
90
80
70
20
20000
FREQUENCY IN CYCLES PER SECOND
Sound Pressure Level in DB re 0.0002 dynes/cm²

Figure A5-41



Effect Of Varying Length Of Clear Vinyl Tubing,
0.190 in. ID x ^ 230 in. O.D. (1 mw input)
2 in.
4 in.
6 in.
8 in.
Sound Pressure Level in DB re 0.0002 dynes/cm²
130
120
110
100
90
80
70
20000
FREQUENCY IN CYCLES PER SECOND

Figure A5-42



Inside diameter 0.285 in.
Inside diameter 0.365 in.
120
110
100
90
Sound Pressure Level In DB re 0.0002 dynes/cm²
20000
FREQUENCY IN CYCLES PER SECOND
Effect Of Varying Diameter Of 6" Length
Clear Vinyl Tubing (1 mw Input)

Figure A5-43

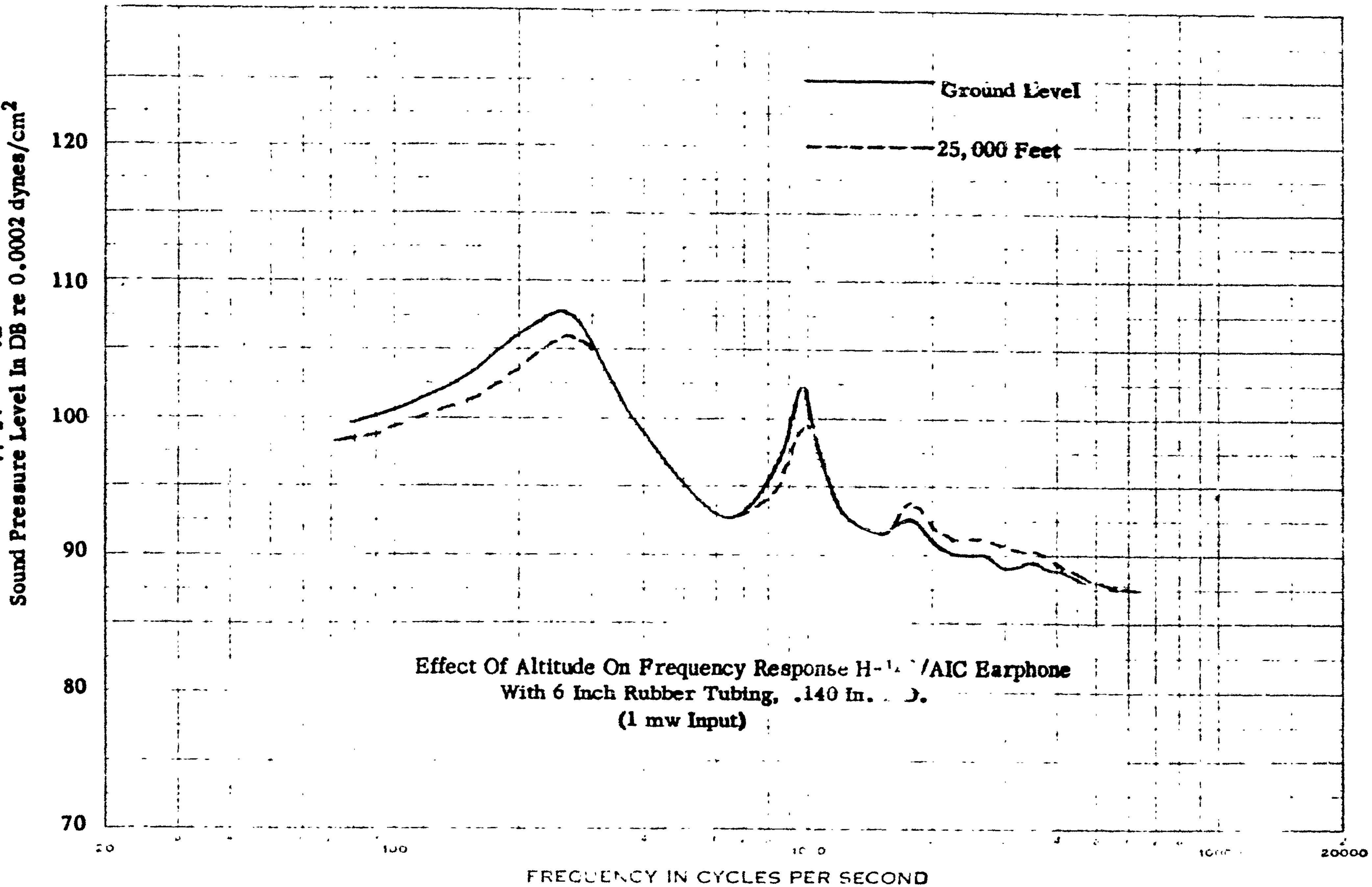

Ground Level
25, 000 Feet
Effect Of Altitude On Frequency Response
/AIC Earphone
With 6 Inch Rubber Tubing, .140 In.
(1 mw Input)
120
110
100
90
80
70
100
20000
FREQUENCY IN CYCLES PER SECOND
Sound Pressure Level In DB re 0.0002 dynes/cm²

Figure 15-44

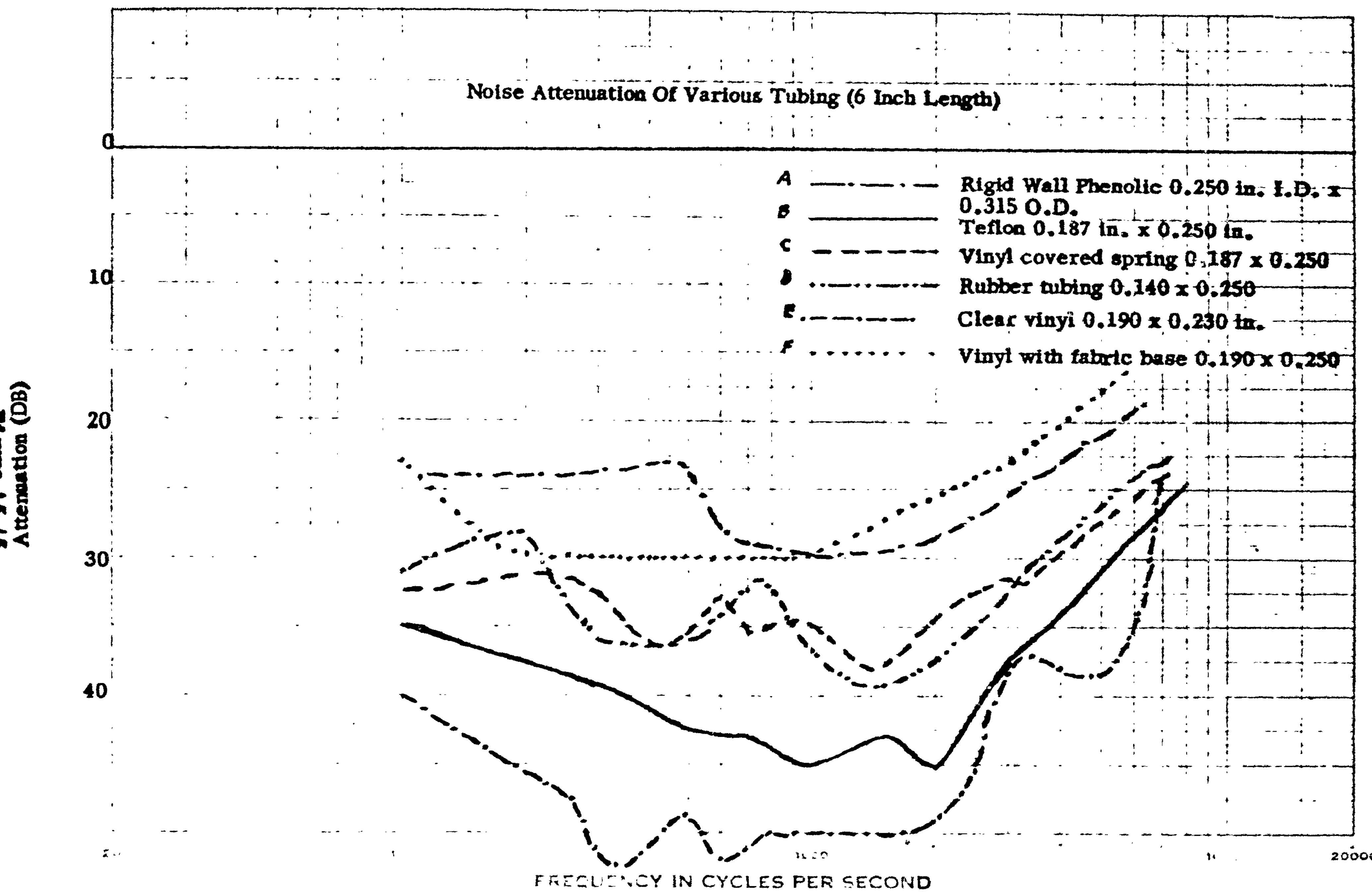

Noise Attenuation Of Various Tubing (6 Inch Length)
A Rigid Wall Phenolic 0.250 in. I.D. x 0.315 O.D.
B Teflon 0.187 in. x 0.250 in.
C Vinyl covered spring 0.187 x 0.250
D Rubber tubing 0.140 x 0.250
E Clear vinyl 0.190 x 0.230 in.
F Vinyl with fabric base 0.190 x 0.250
0
10
20
30
40
Attenuation (DB)
20000
FREQUENCY IN CYCLES PER SECOND

Figure 15-45

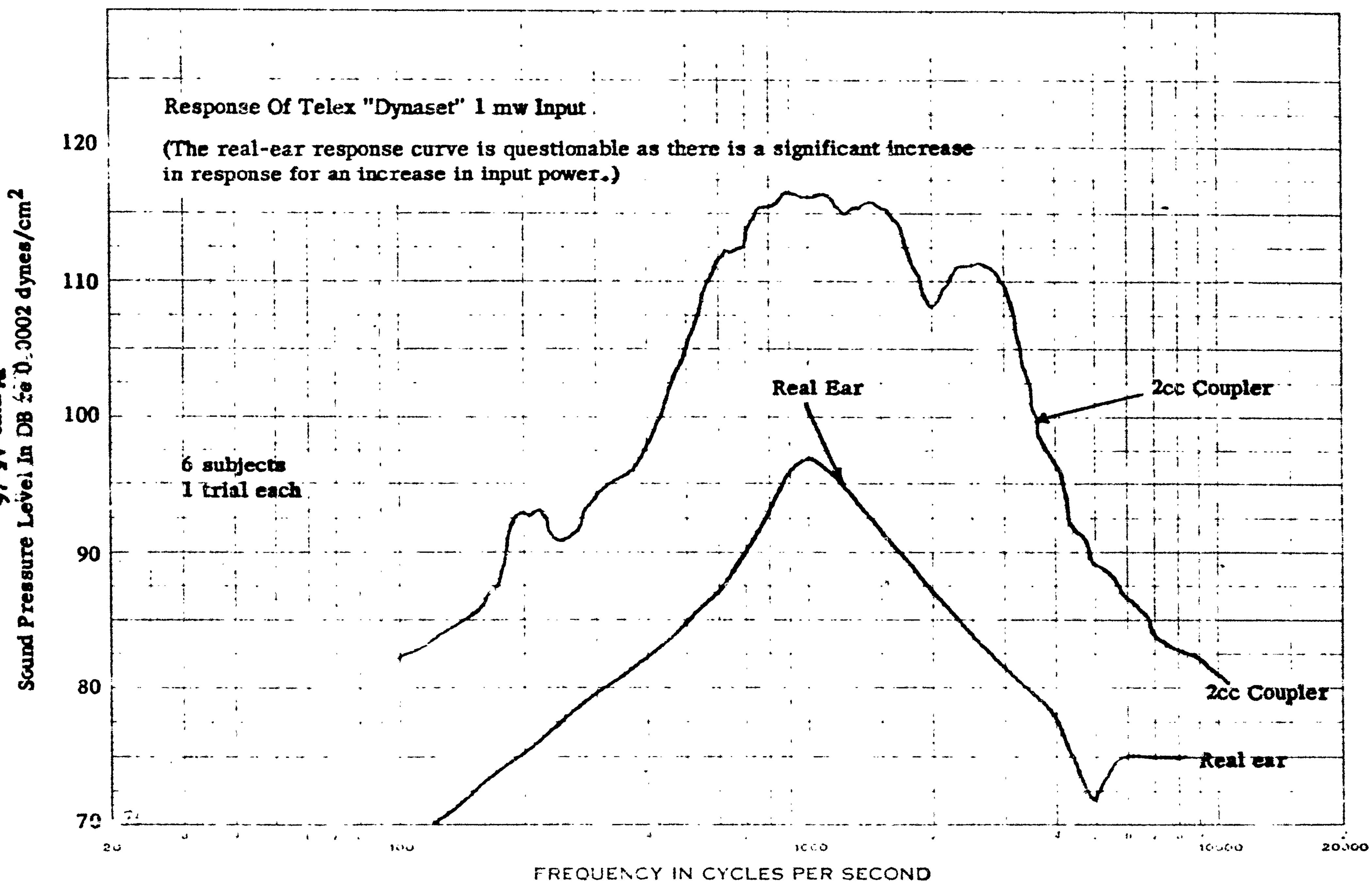

Response Of Telex "Dynaset" 1 mw Input
(The real-ear response curve is questionable as there is a significant increase in response for an increase in input power.)
6 subjects
1 trial each
Real Ear
2cc Coupler
2cc Coupler
Real ear
120
110
100
90
80
70
20
100
1000
10000
20000
FREQUENCY IN CYCLES PER SECOND
Sound Pressure Level in DB re 0.0002 dynes/cm²

Figure A5-46

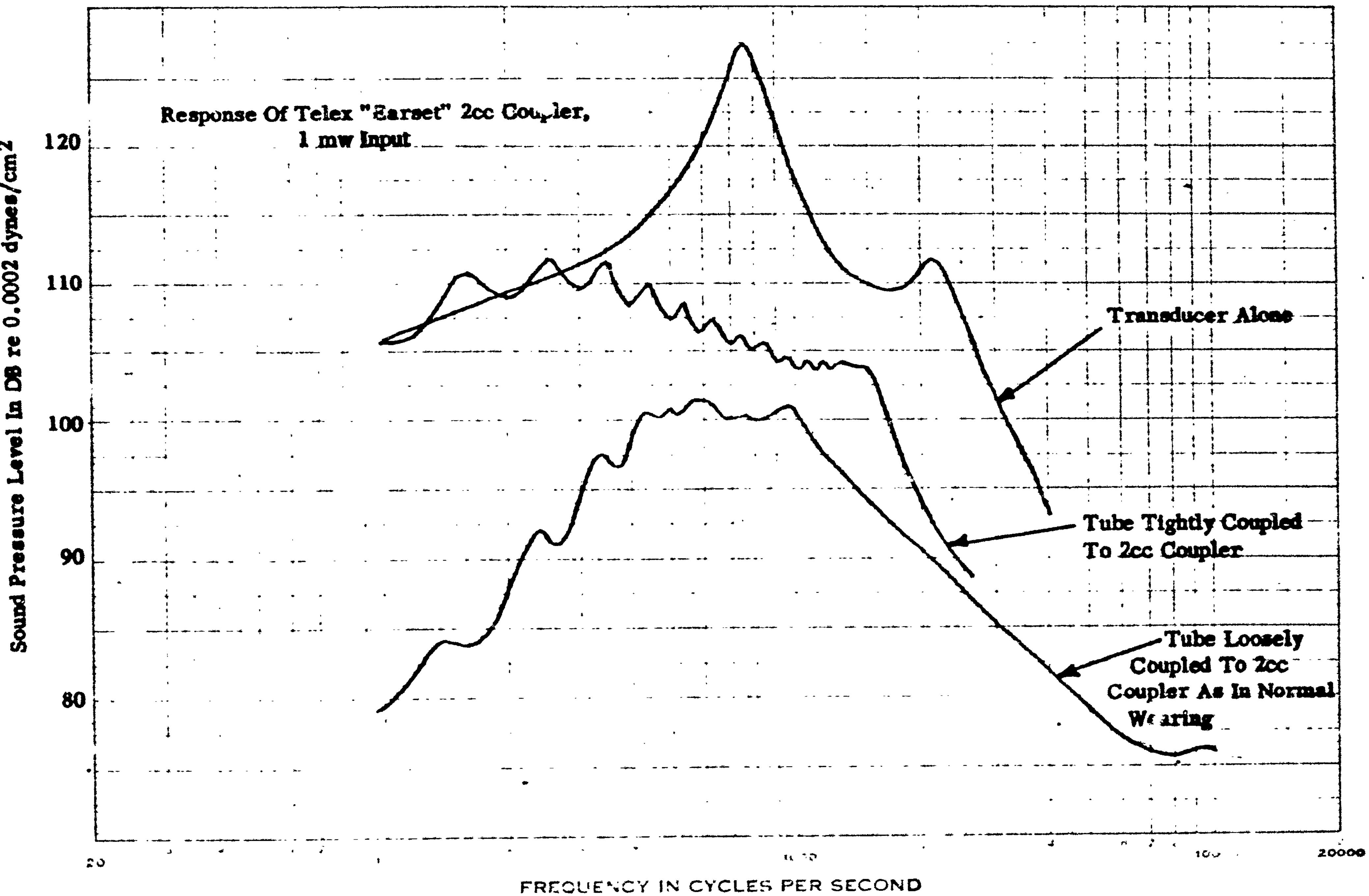
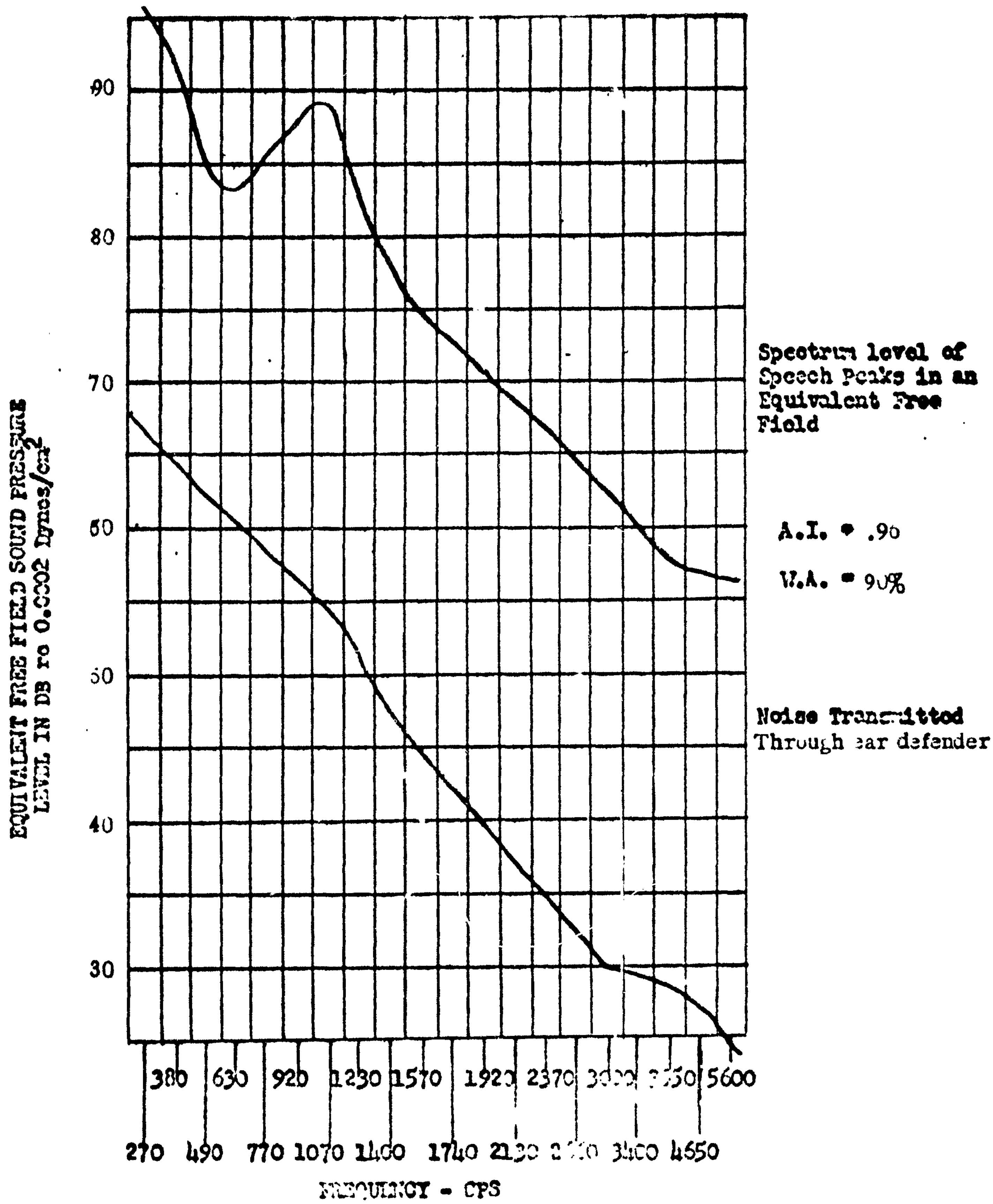
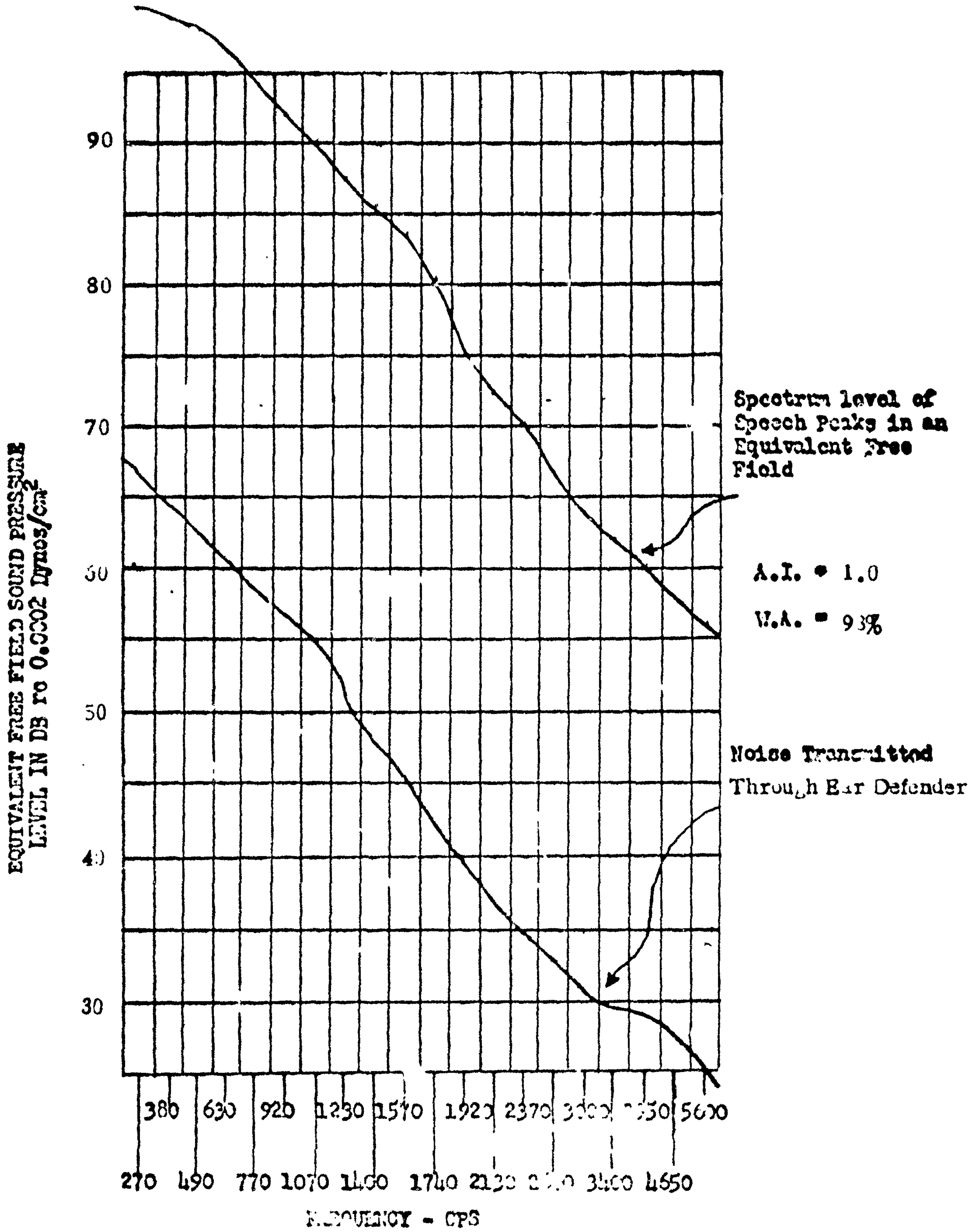

Response Of Telex "Earset" 2cc Coupler,
1 mw Input
Transducer Alone
Tube Tightly Coupled
To 2cc Coupler
Tube Loosely
Coupled To 2cc
Coupler As In Normal
Wearing
120
110
100
90
80
20
20000
FREQUENCY IN CYCLES PER SECOND
Sound Pressure Level in DB re 0.0002 dynes/cm2

Figure A5-47

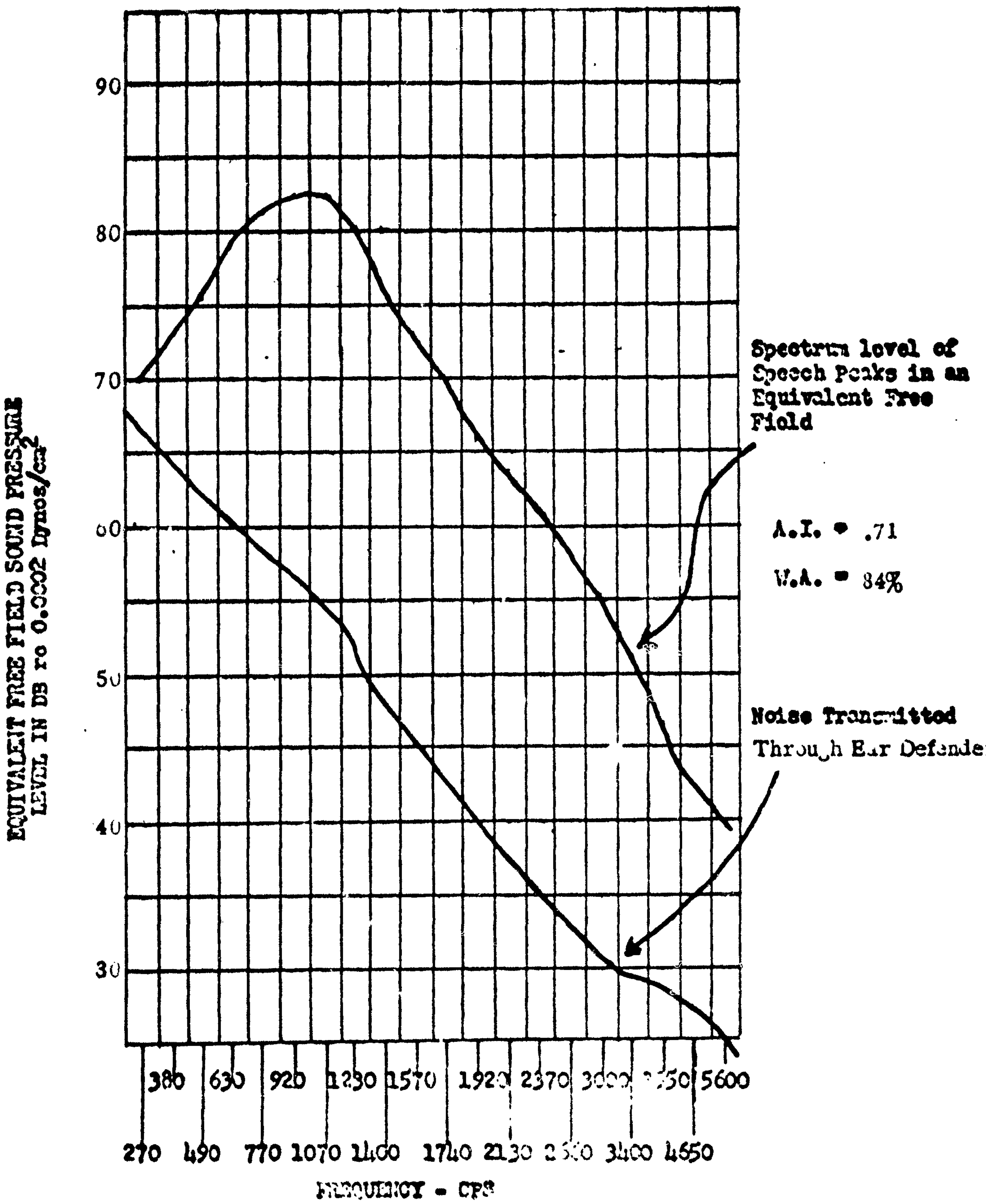


ARTICULATION INDEX COMPUTATION CHART
FOR
H-143 Earphone Coupled To The Ear Via 6" Rubber Tube In An MSA Ear Defender
200 mw, no clipping

Figure A5-48



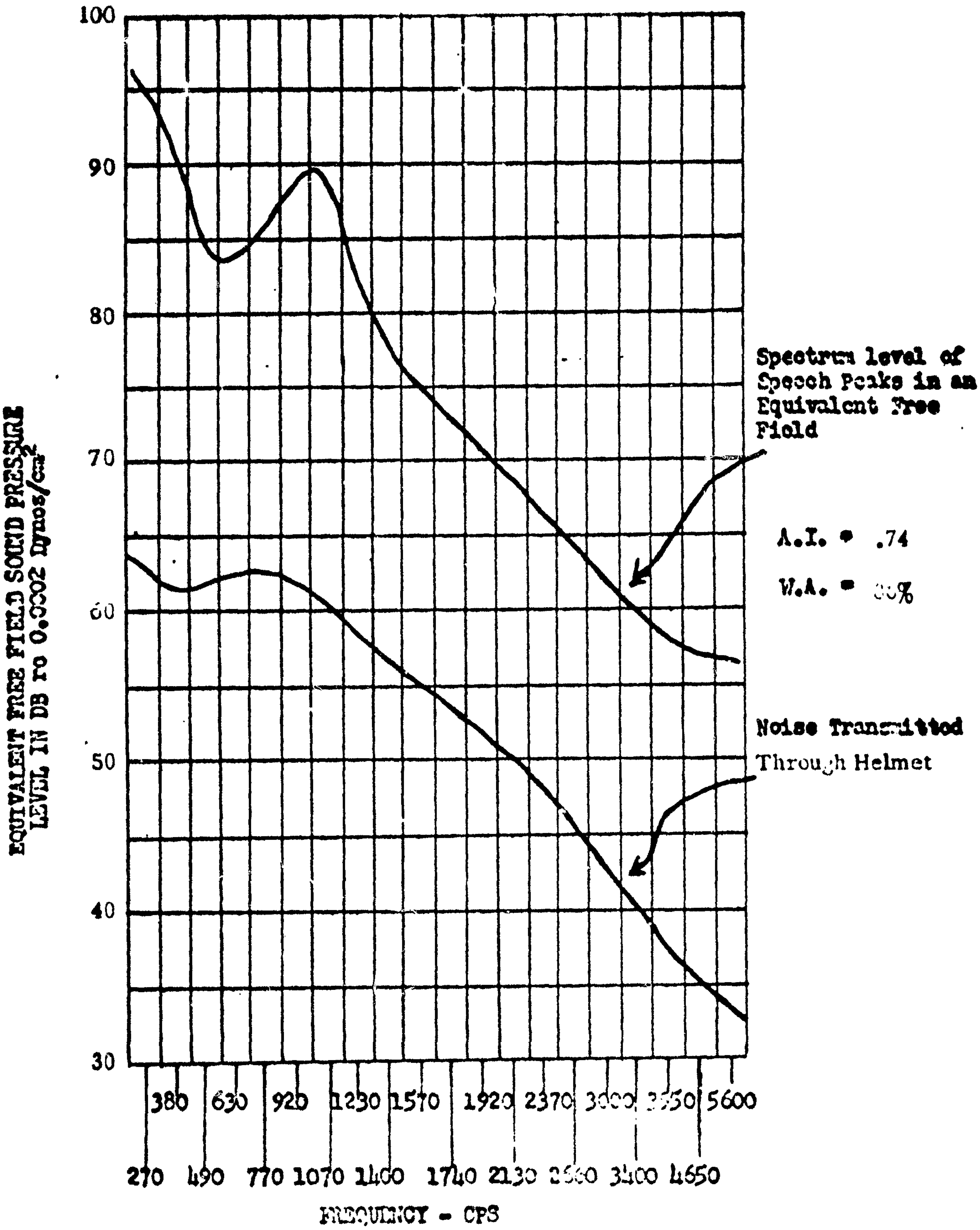
ARTICULATION INDEX COMPUTATION CHART
FOR
Telex "Earset" Coupled To The Ears Via MSA Ear Defender
200 mw, no clipping
Figure A5-49



ARTICULATION INDEX COMPUTATION CHART
FOR

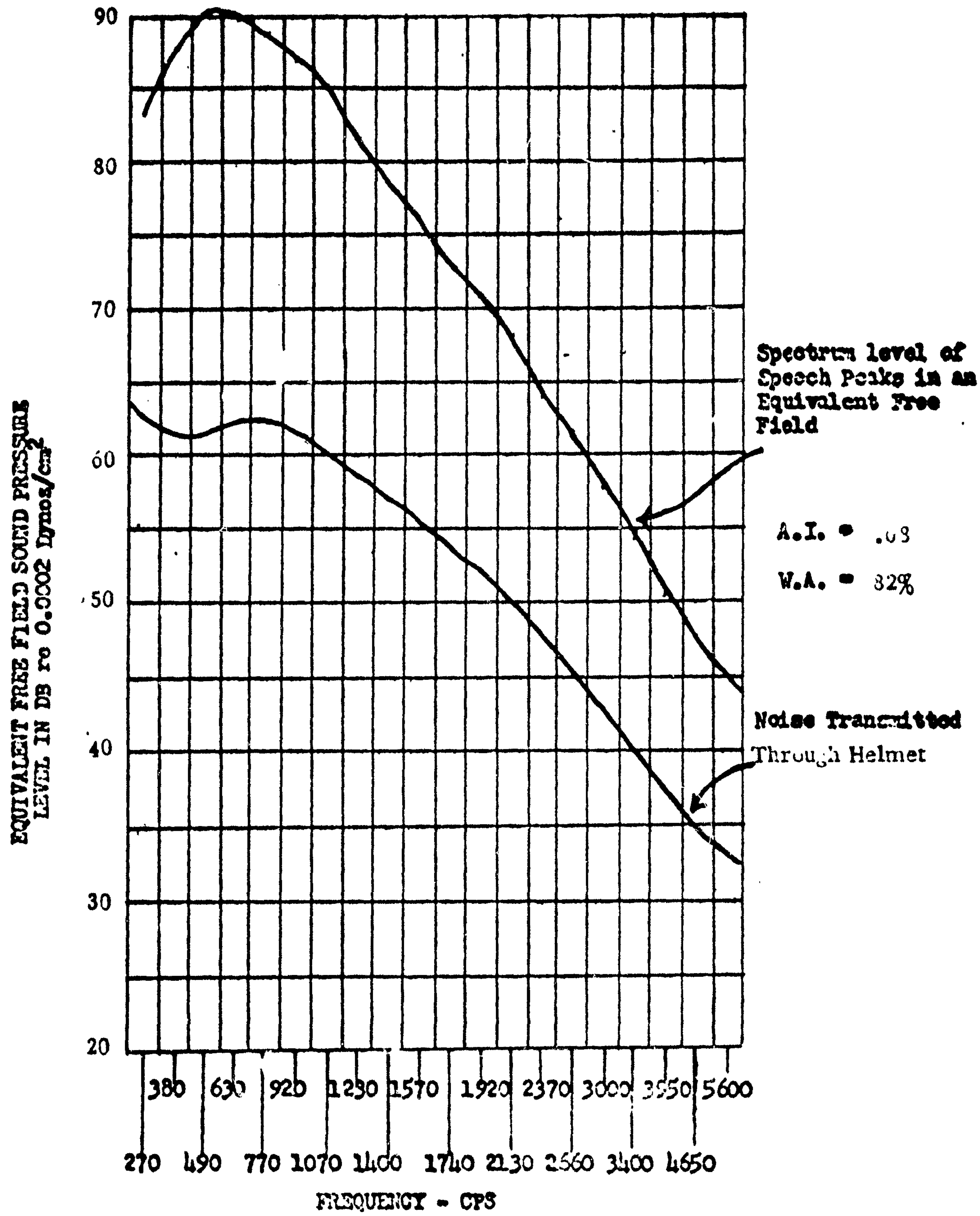
Telex "Dynaset" Coupled To The Ears Via MAS Ear Defender
(200 mw, no clipping)

Figure A5-50



ARTICULATION INDEX COMPUTATION CHART
FOR
H-143 Earphone Coupled To The Ear Via ." Rubber Tube In An Experimental Plastic Helmet
(20 mw, no clipping)

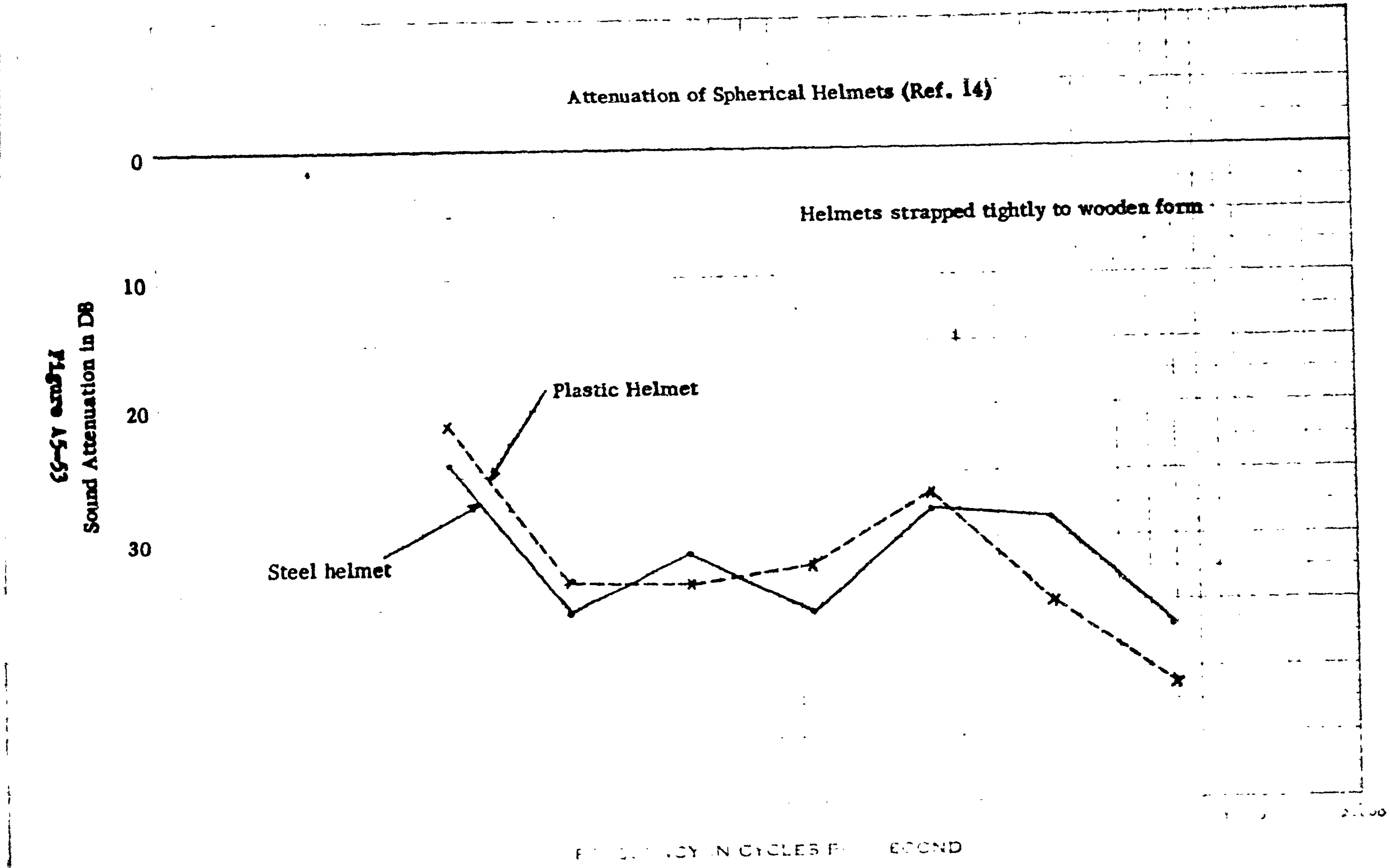
Figure A5-51



ARTICULATION INDEX COMPUTATION CHART
FOR

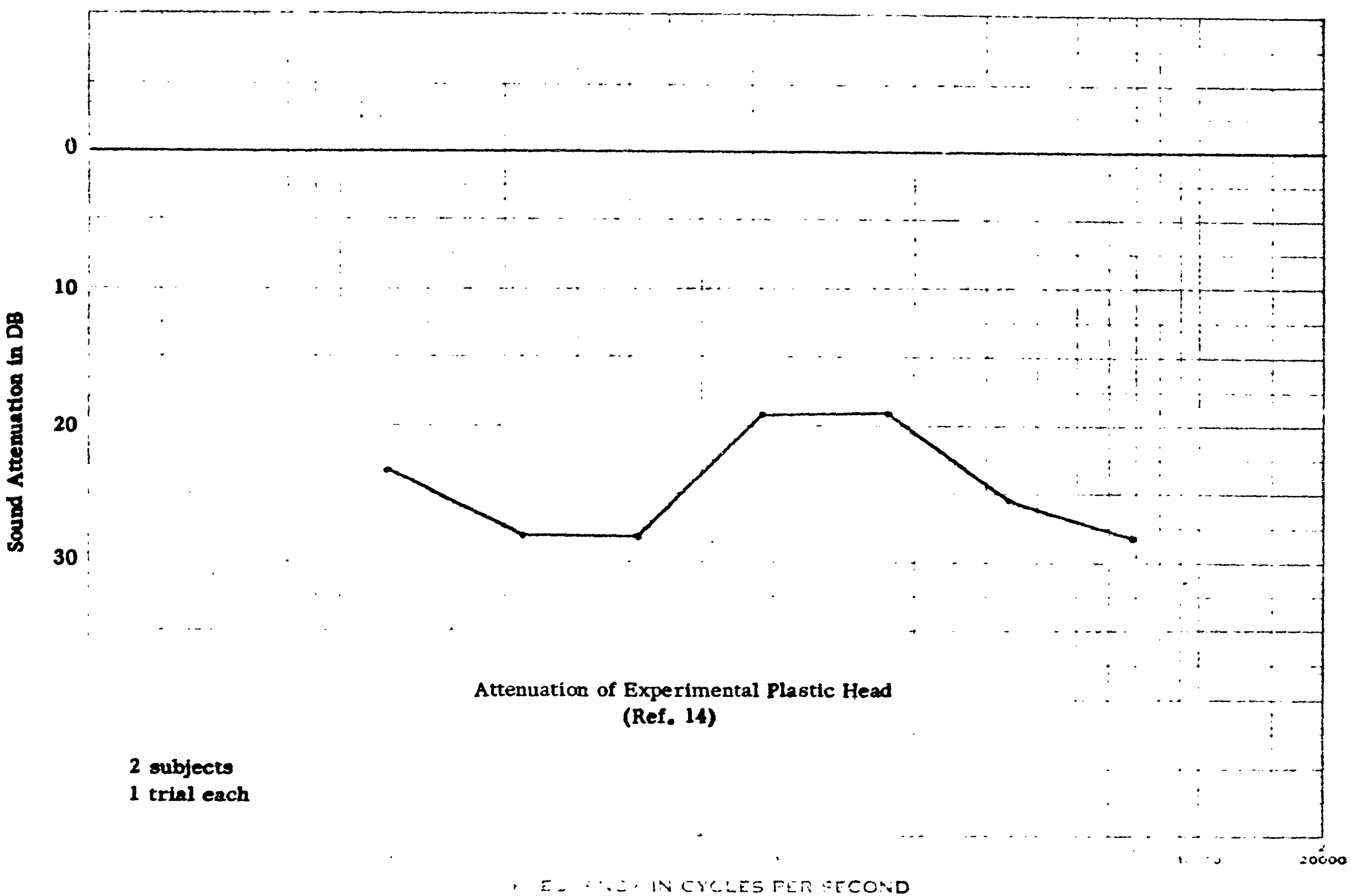
Telex "Earset" Loosely Coupled To The Ears Using The Experimental Plastic Helmet
(250mw, no clipping)

Figure A5-52



Attenuation of Spherical Helmets (Ref. 14)
Helmets strapped tightly to wooden form
Plastic Helmet
Steel helmet
0
10
20
30
Sound Attenuation in DB

Figure A5-53



0
10
20
30
Sound Attenuation in DB
Attenuation of Experimental Plastic Head
(Ref. 14)
2 subjects
1 trial each
20000
IN CYCLES PER SECOND

Figure A5-54

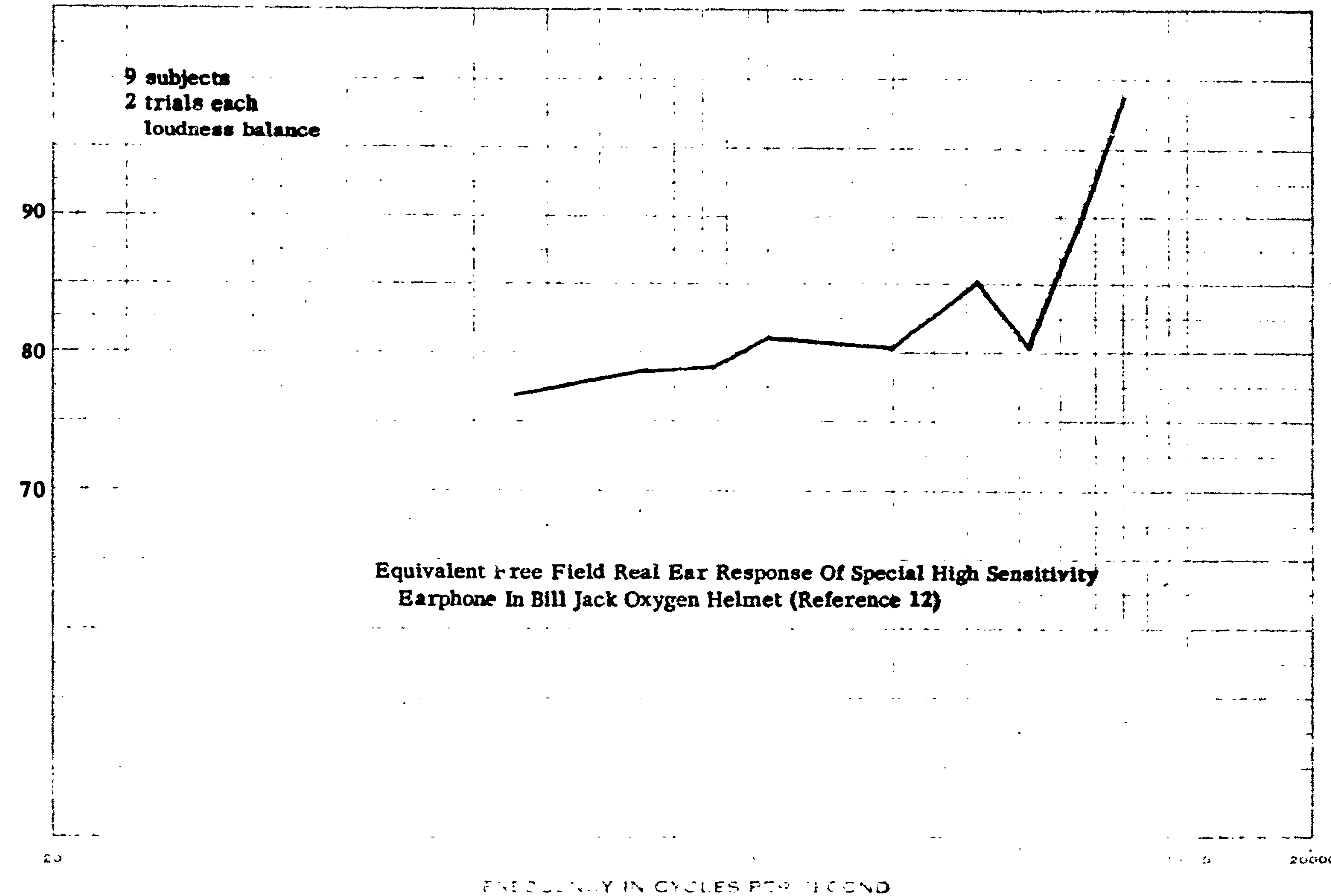
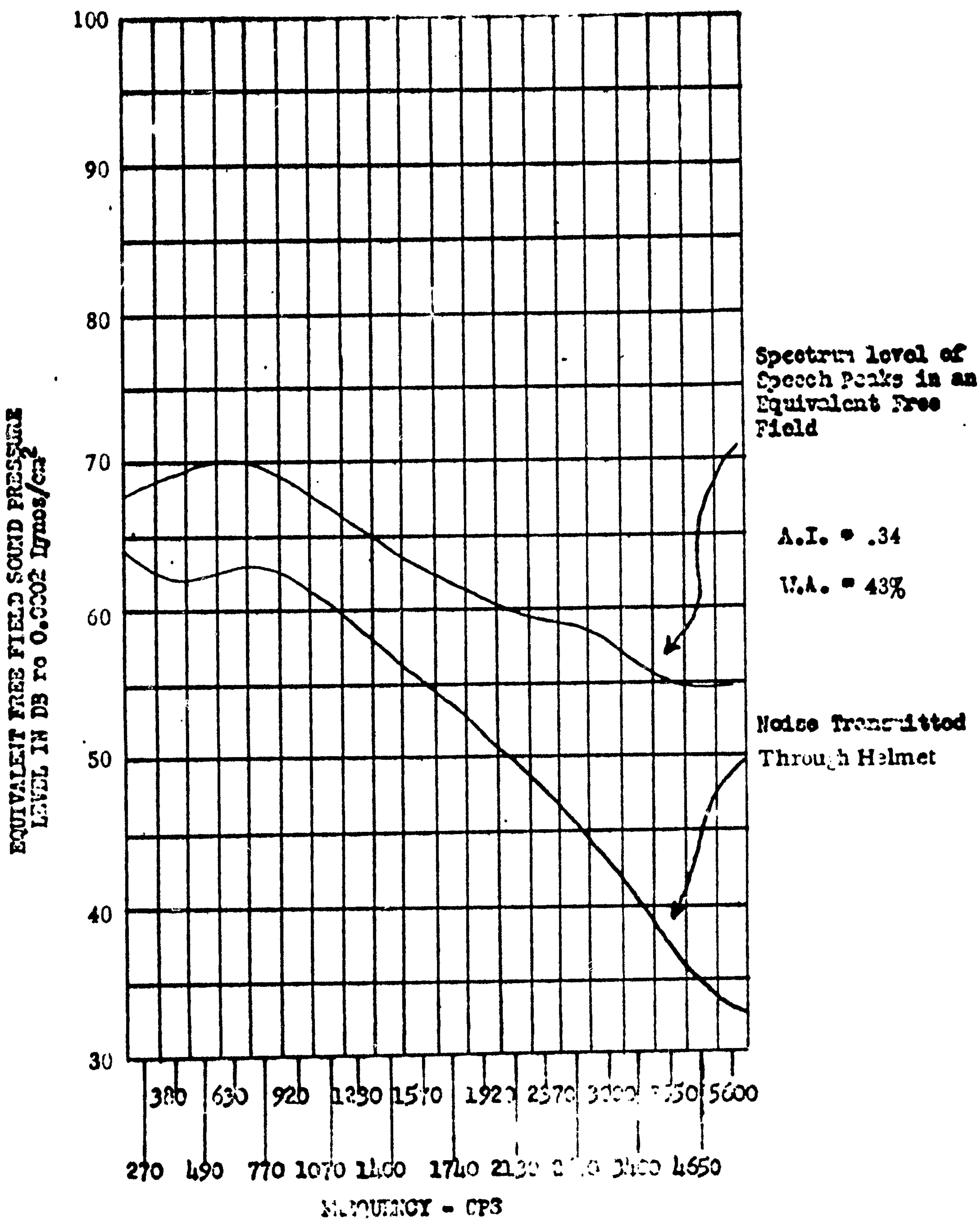


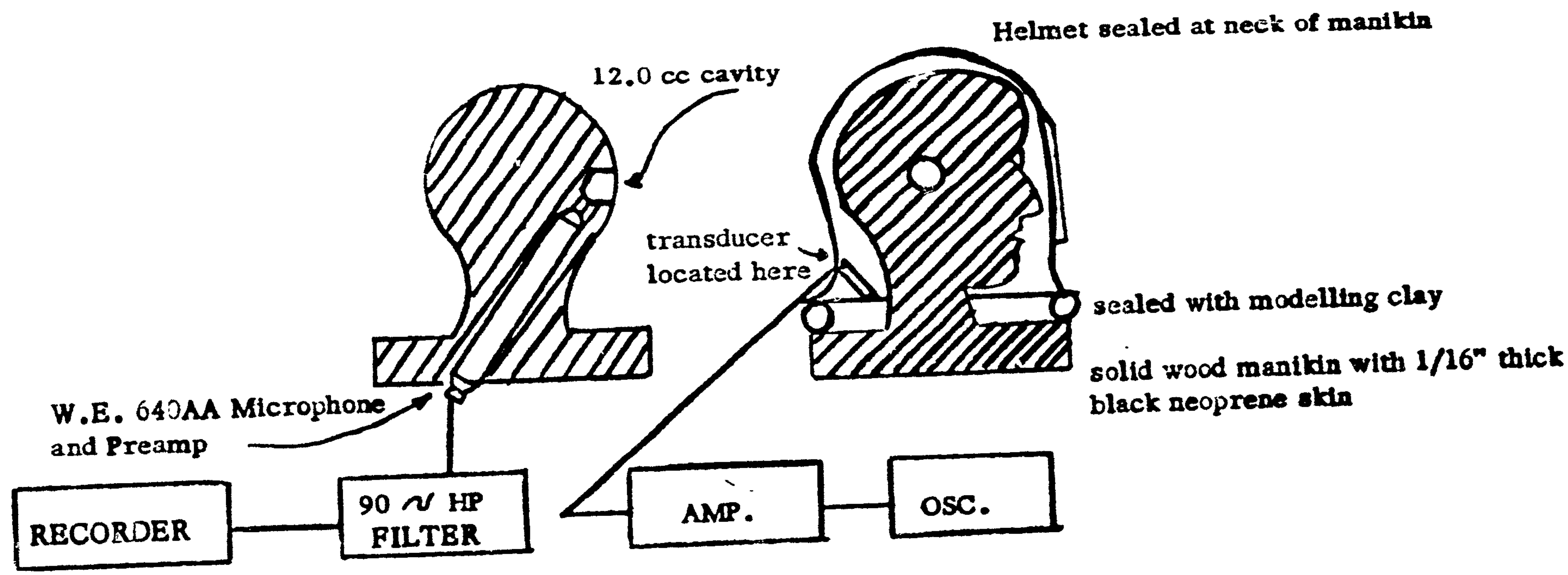
Figure 45-55
Equivalent Free Field Sound Pressure Level in DB re 0.0002 dynes/cm^2 for 1mw Power Available



ARTICULATION INDEX COMPUTATION CHART FOR

Hi Sensitivity Earphone In Experimental Helmet (Loosely Coupled)
(20 mw, no clipping)

Figure A5-56



Means For Measuring Frequency Response Of Various Transducers
Exciting A Cavity Form By A Helmet On A Solid Wood Manikin

Figure A5-57

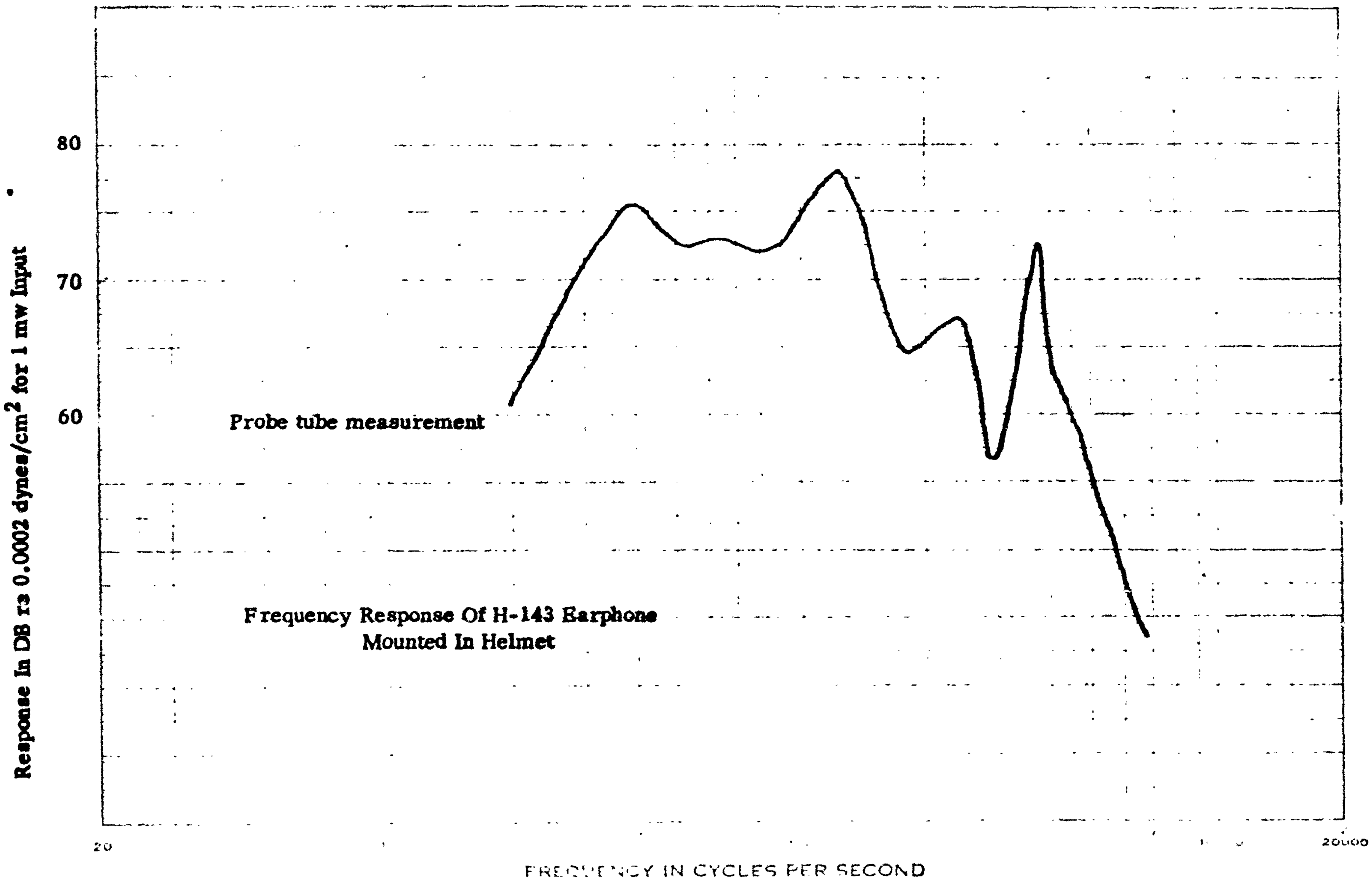

80
70
60
Response In DB re 0.0002 dynes/cm² for 1 mw Input
Probe tube measurement
Frequency Response Of H-143 Earphone
Mounted In Helmet
20
20000
FREQUENCY IN CYCLES PER SECOND

Figure A5-58

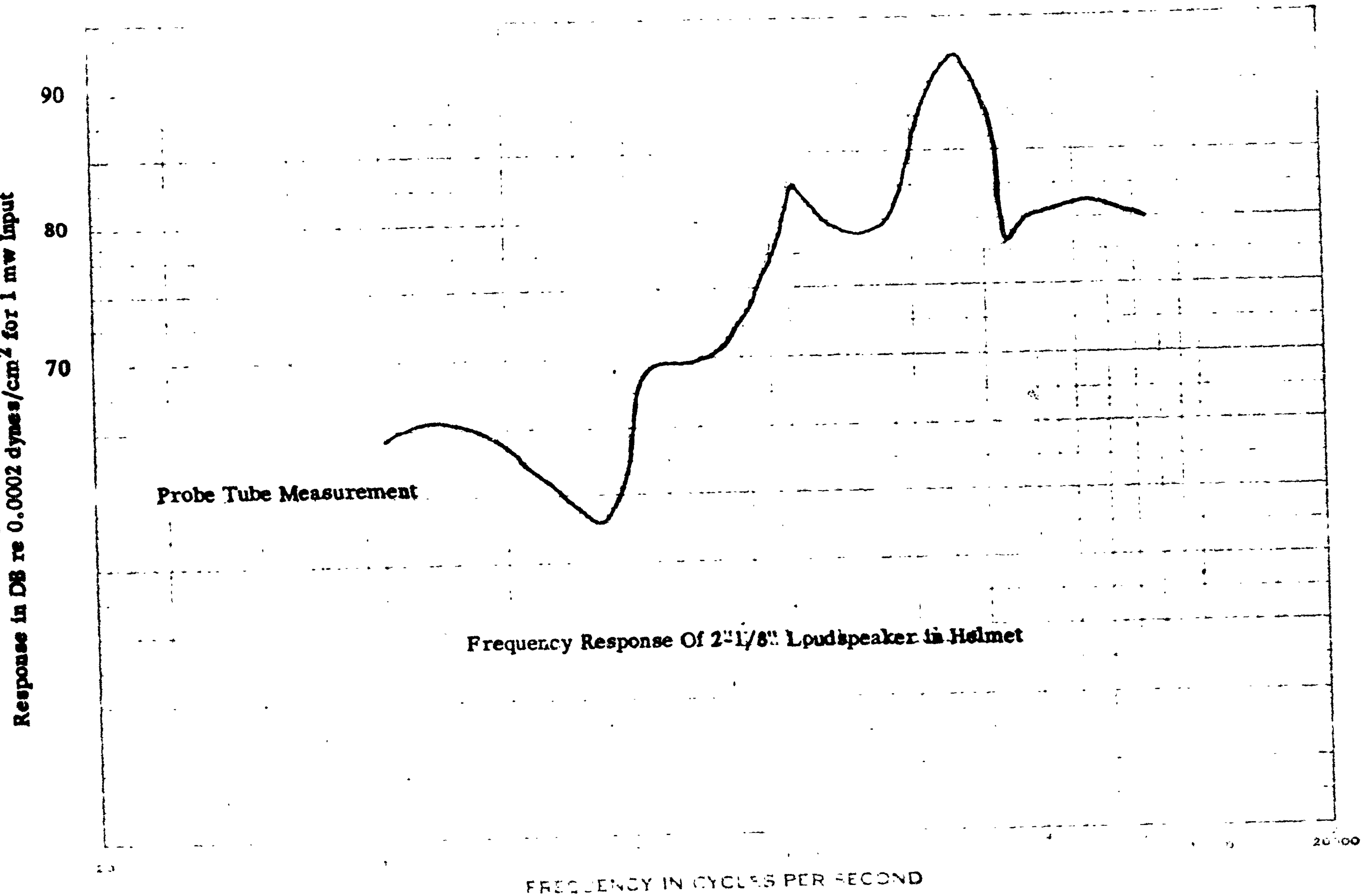
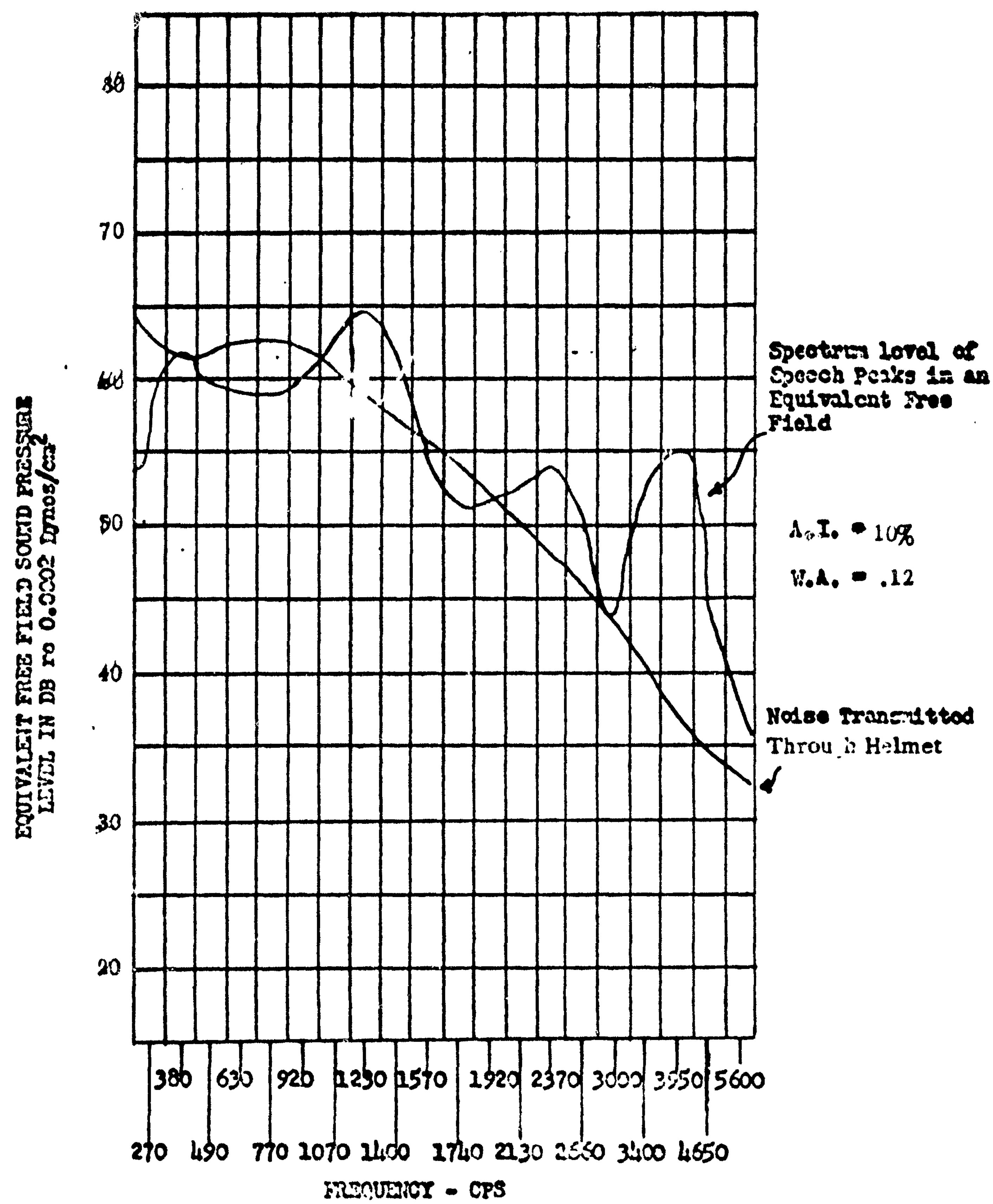

Response in DB re 0.0002 dynes/cm² for 1 mw Input
90
80
70
Probe Tube Measurement
Frequency Response Of 2-1/8" Loudspeaker in Helmet
20,000
FREQUENCY IN CYCLES PER SECOND

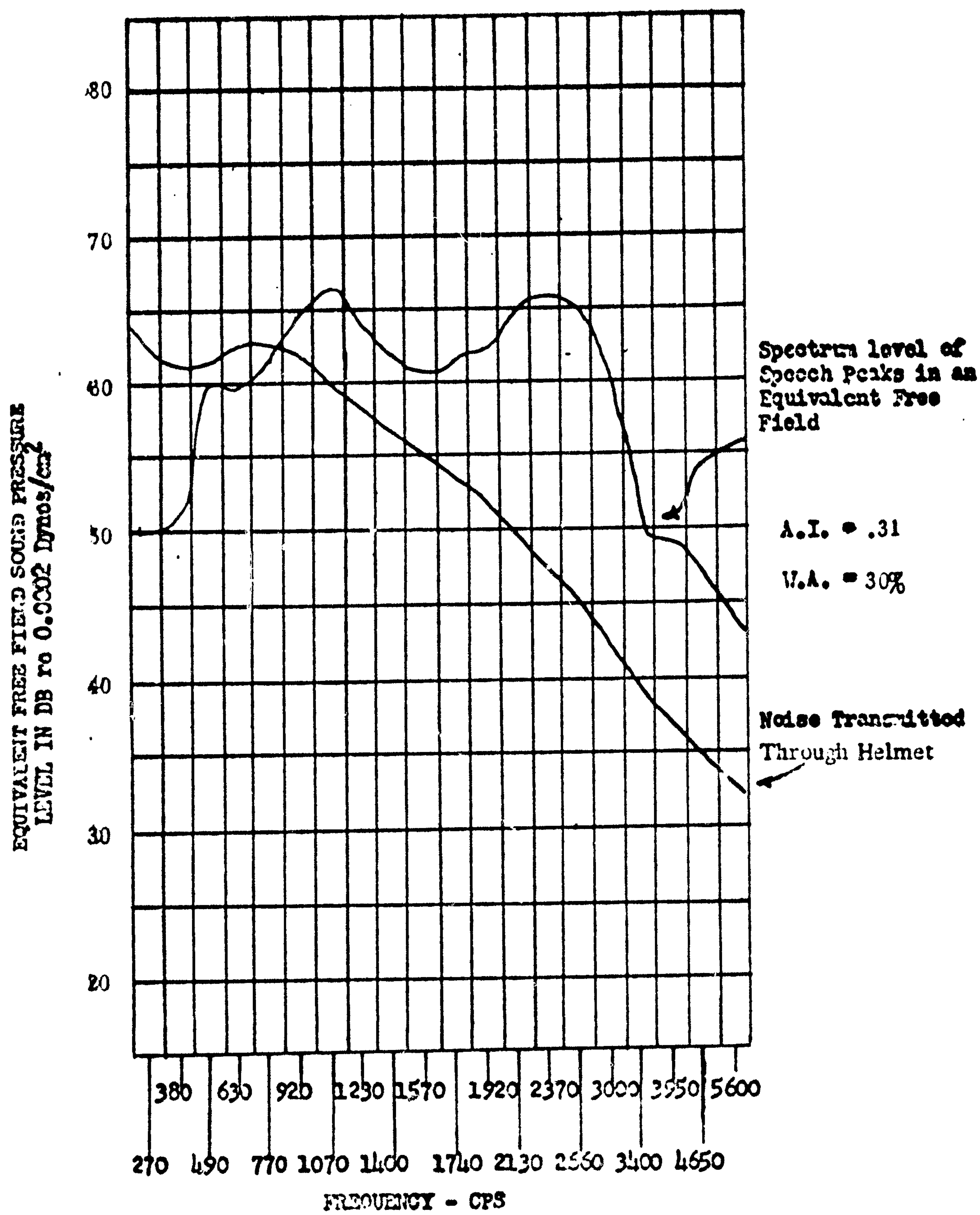
Figure A5-59



ARTICULATION INDEX COMPUTATION CHART
FOR

H-143 Earphone In Experimental Plastic Helmet
(200 mw, no clipping)

Figure A5-60



ARTICULATION INDEX COMPUTATION CHART
FOR

2-1/8" Loudspeaker in Experimental Plastic Helmet
(200 mw, no clipping)

Figure A5-61

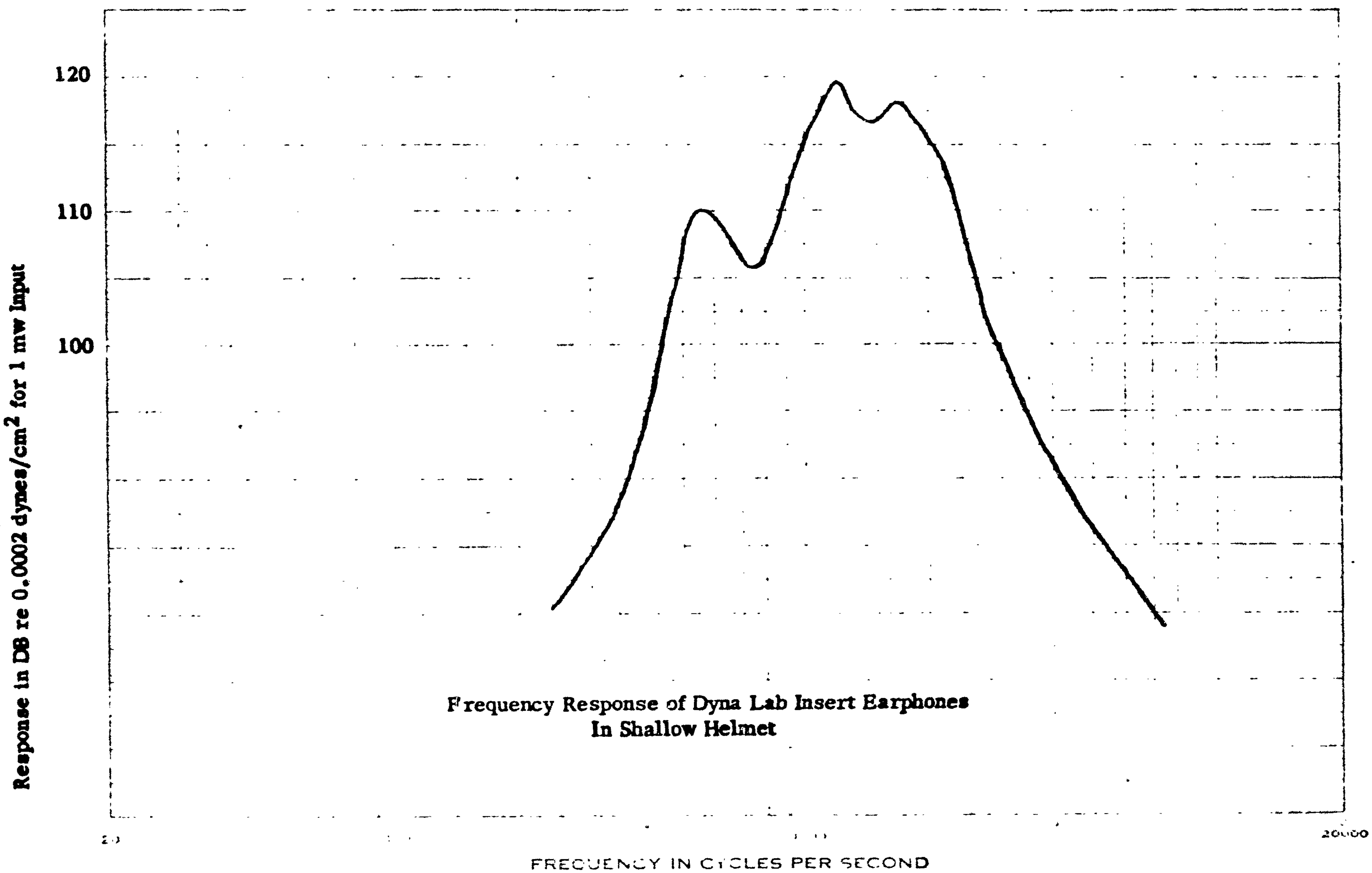
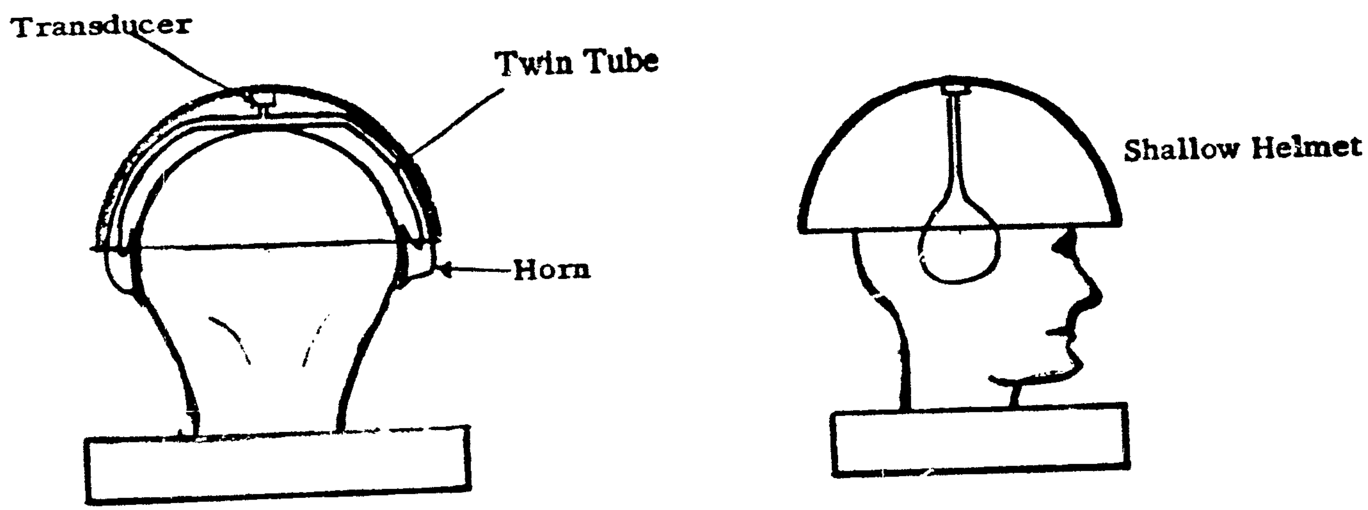

120
110
100
Response in DB re 0.0002 dynes/cm² for 1 mw Input
Frequency Response of Dyna Lab Insert Earphones
In Shallow Helmet
20000
FRECUENCY IN CYCLES PER SECOND

Figure A5-62



Arrangement Showing A Transducer Mounted At The Top Of A Shallow Helmet

Figure A5-63

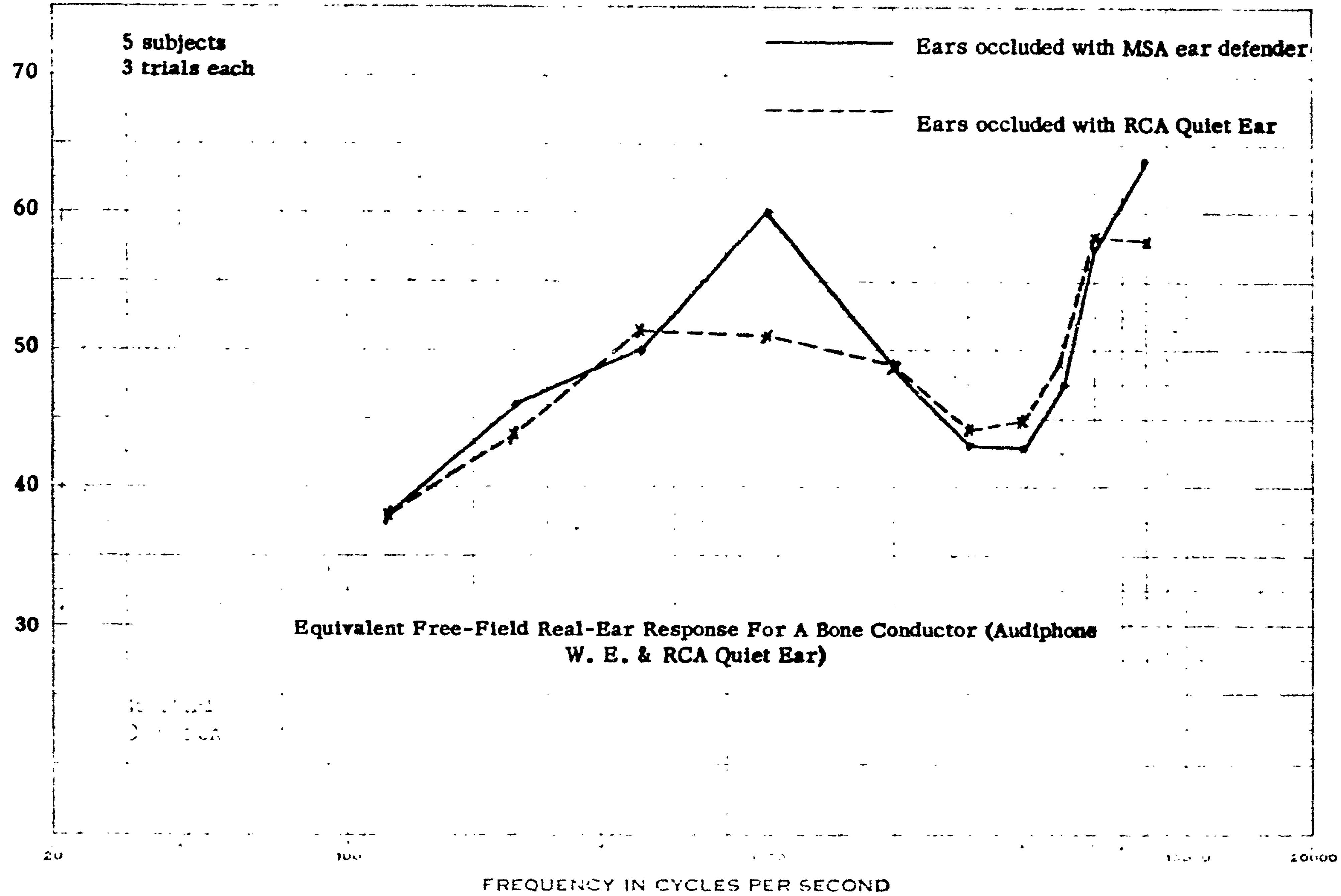
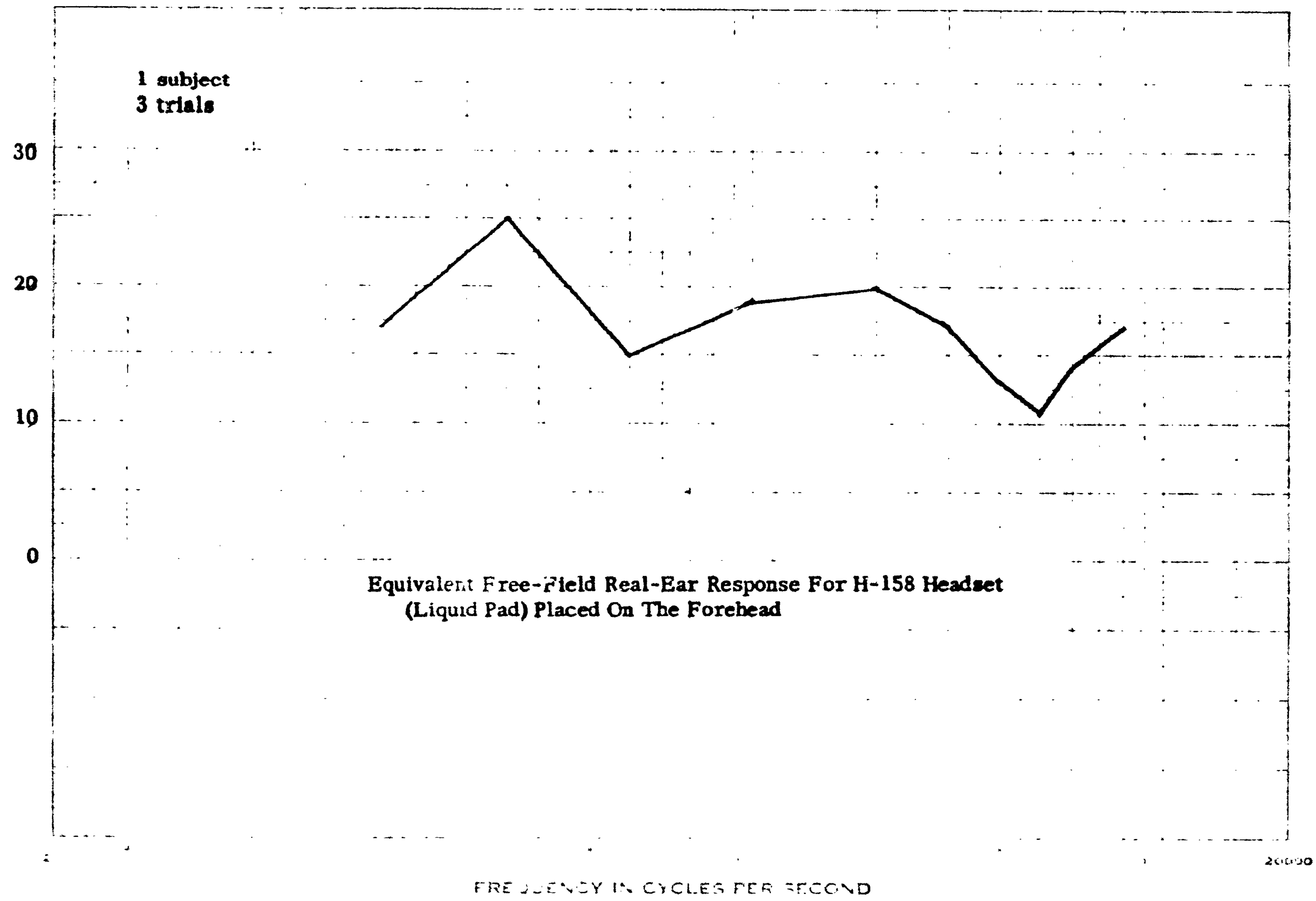


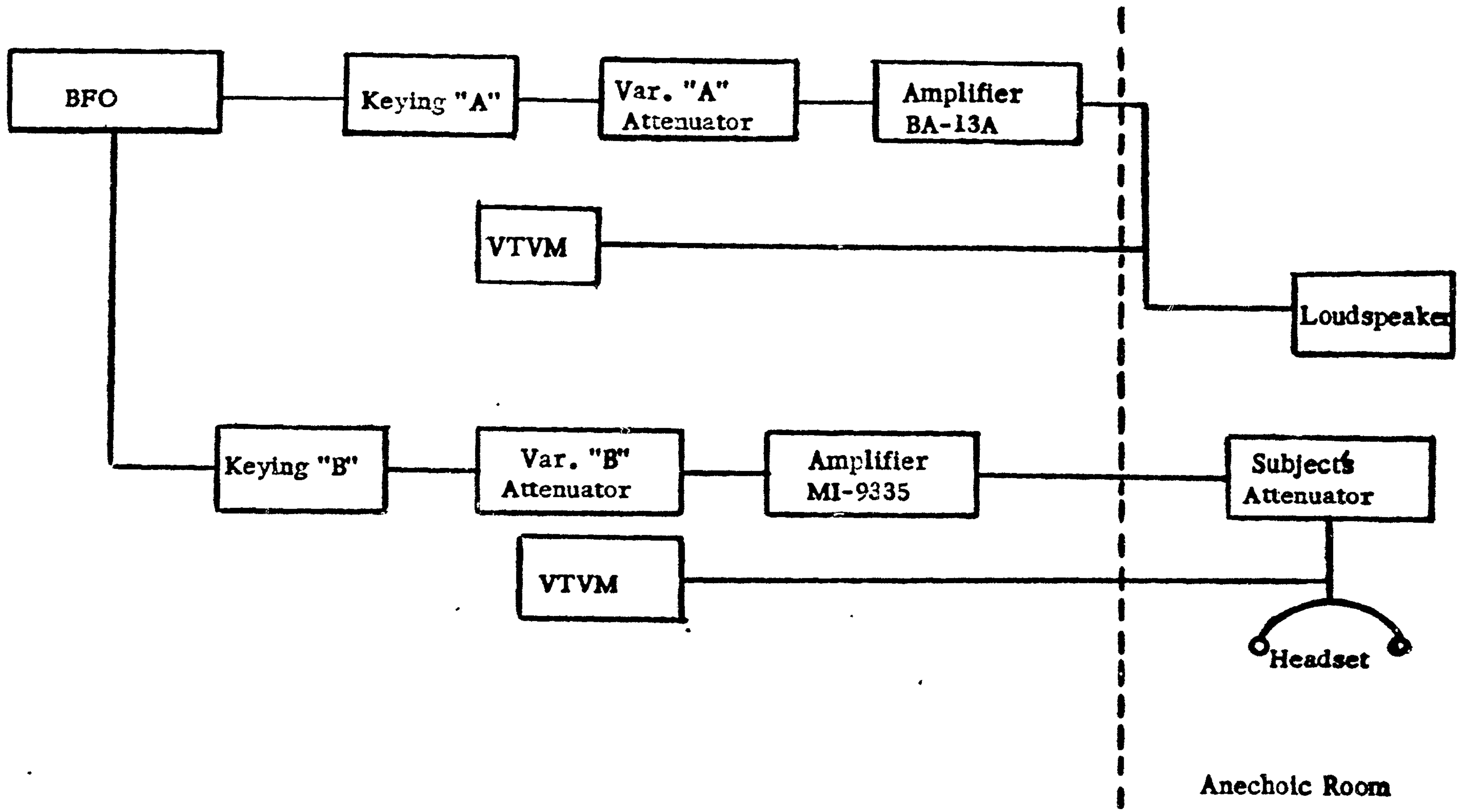
Figure A5-64

Equivalent Free-Field Sound Pressure Level in DB re 0.0002 dynes/cm² for
1 mw Power Available

Figure A5-65

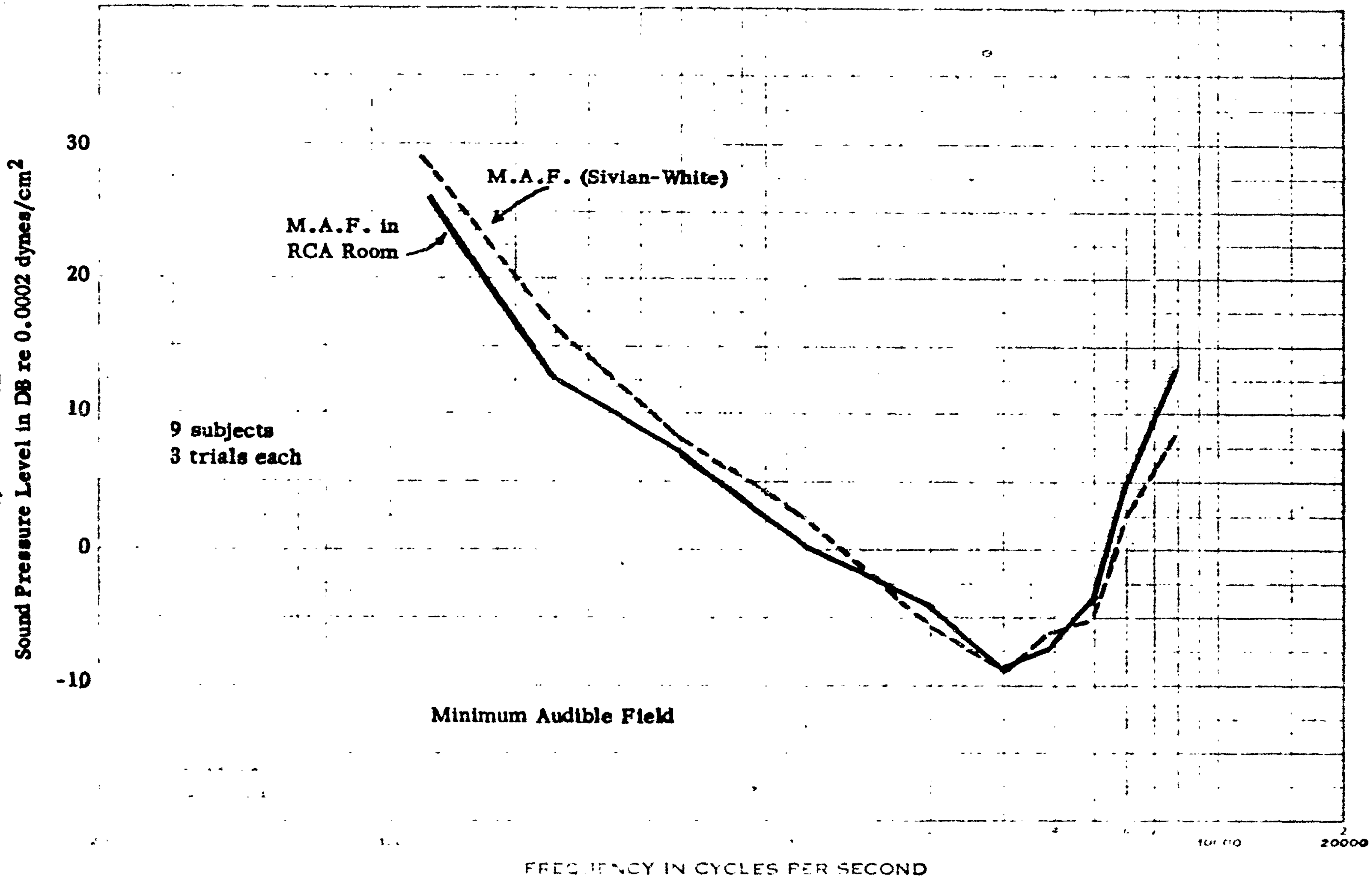


Equivalent Free Field Sound Pressure Level in DB re 0.0002 dynes/cm² For
1 mw Power Available



Block Diagram For Making Real-Ear Response Measurements
By Loudness Balance

Figure A5-66



30
20
10
0
-10
Sound Pressure Level in DB re 0.0002 dynes/cm2
M.A.F. (Sivian-White)
M.A.F. in
RCA Room
9 subjects
3 trials each
Minimum Audible Field
20000
FREQUENCY IN CYCLES PER SECOND

Figure A5-67

FORM HL-4
1-57

GRASON-STADLER COMPANY

FOR USE WITH AUDIOMETER TYPE E-800

SEX ______ AGE ______ No. ______

NAME ______

RED: RIGHT
BLUE: LEFT

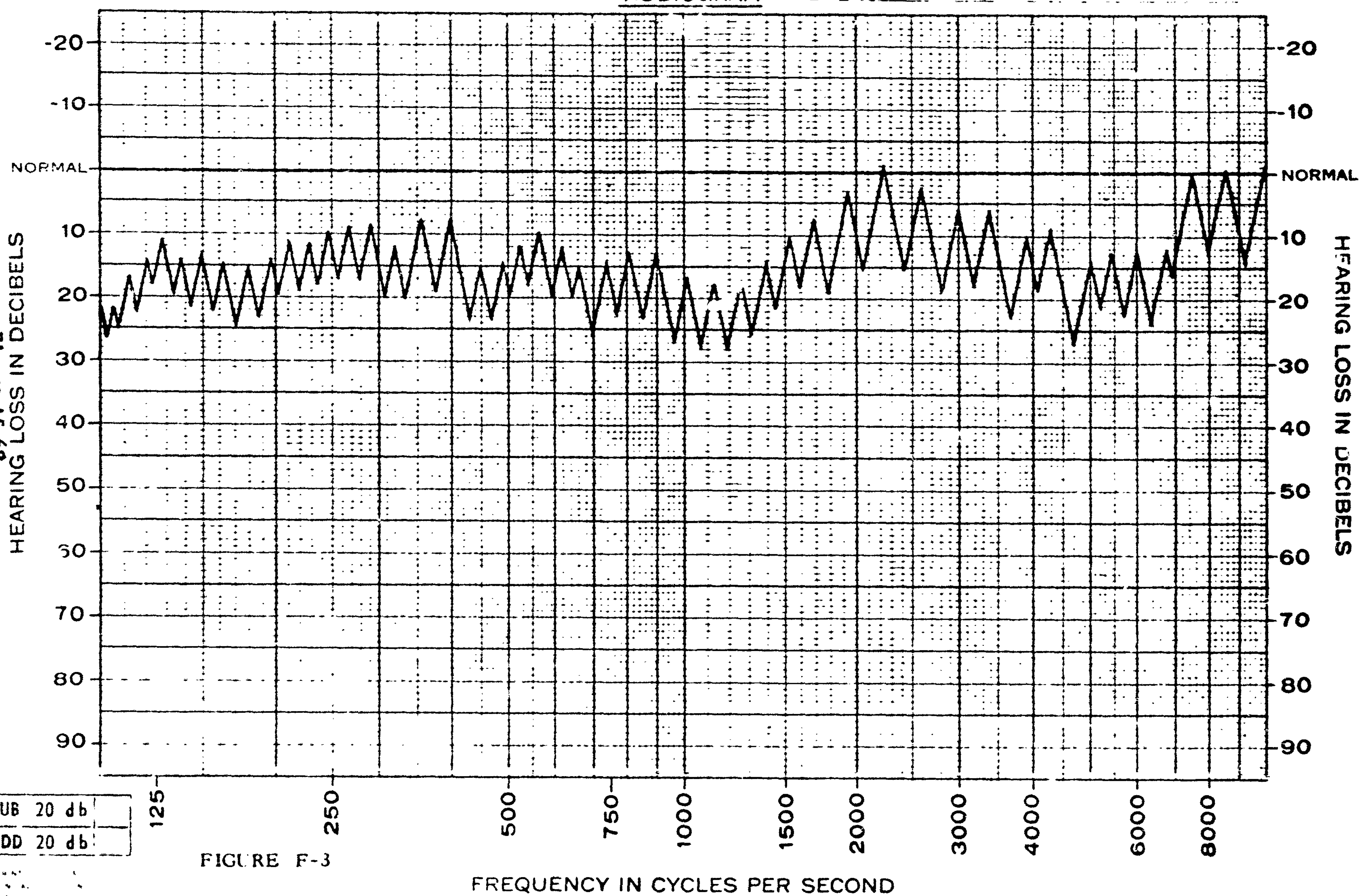
MASKING ______ DB
MASKING ______ DB.

grason stadler

AUDIOGRAM

DATE ______ TIME ______

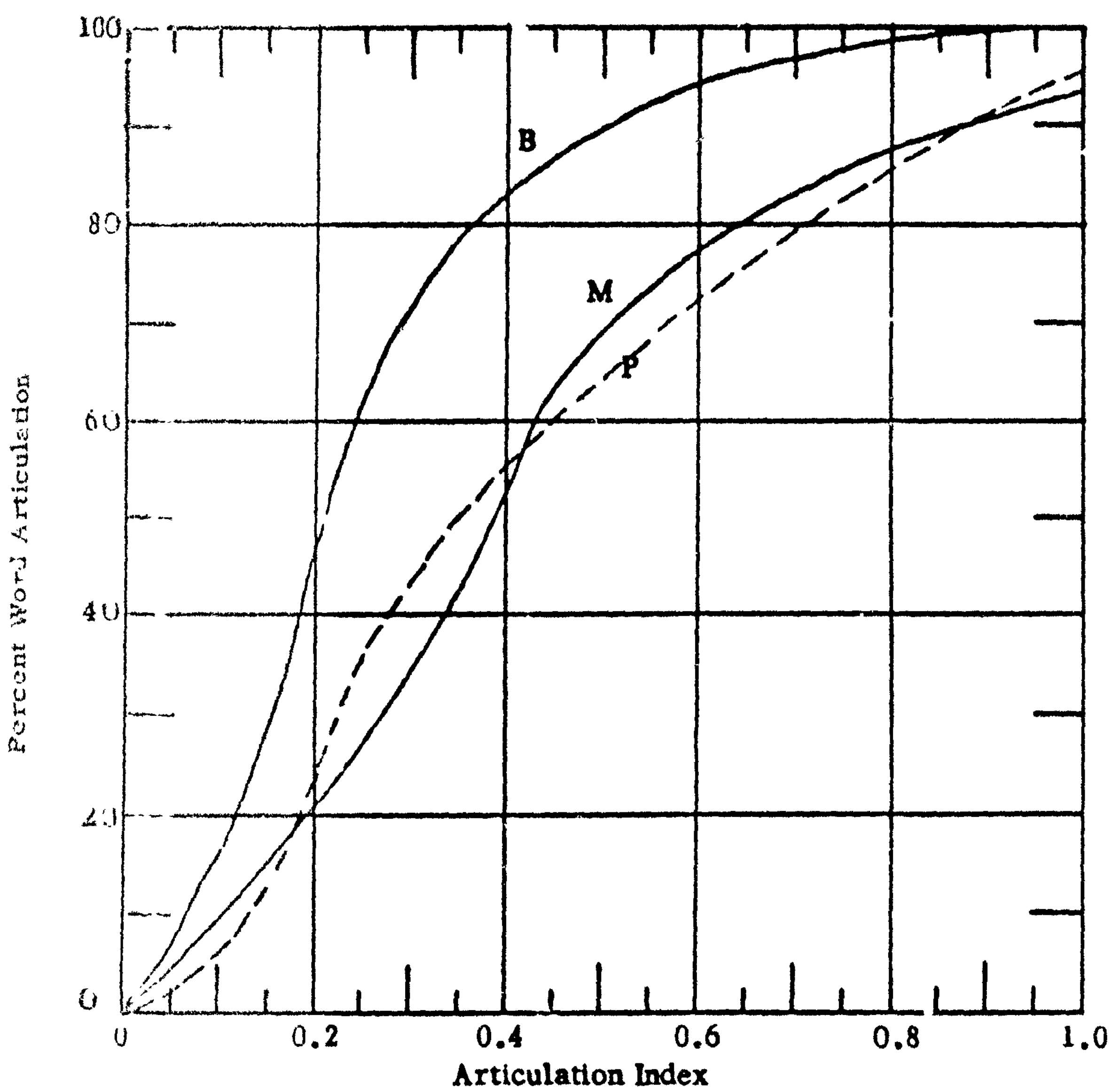
BY ______



SUB 20 db
ADD 20 db

FIGURE F-3

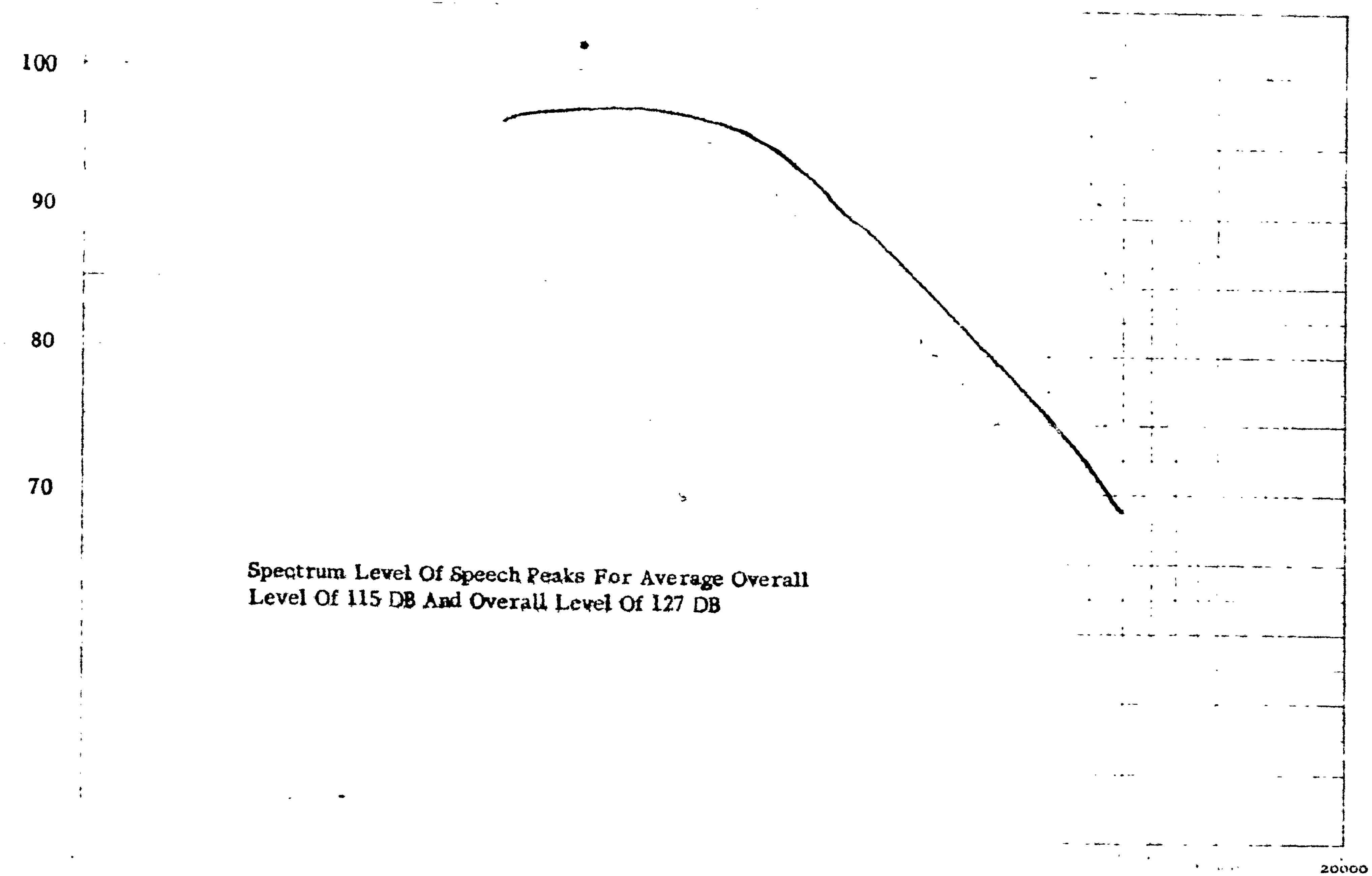
Figure A5-68



Comparison Of Relation Between Word Articulation And Articulation Index Reported By Different Experimenters

B - Beranek - Reference 5
P - Pollack - Reference 7
M - Martin - Reference 6

Figure A5-69



Spectrum Level - DB/Cycle re 0.0002 dynes/cm²

Figure A5-70

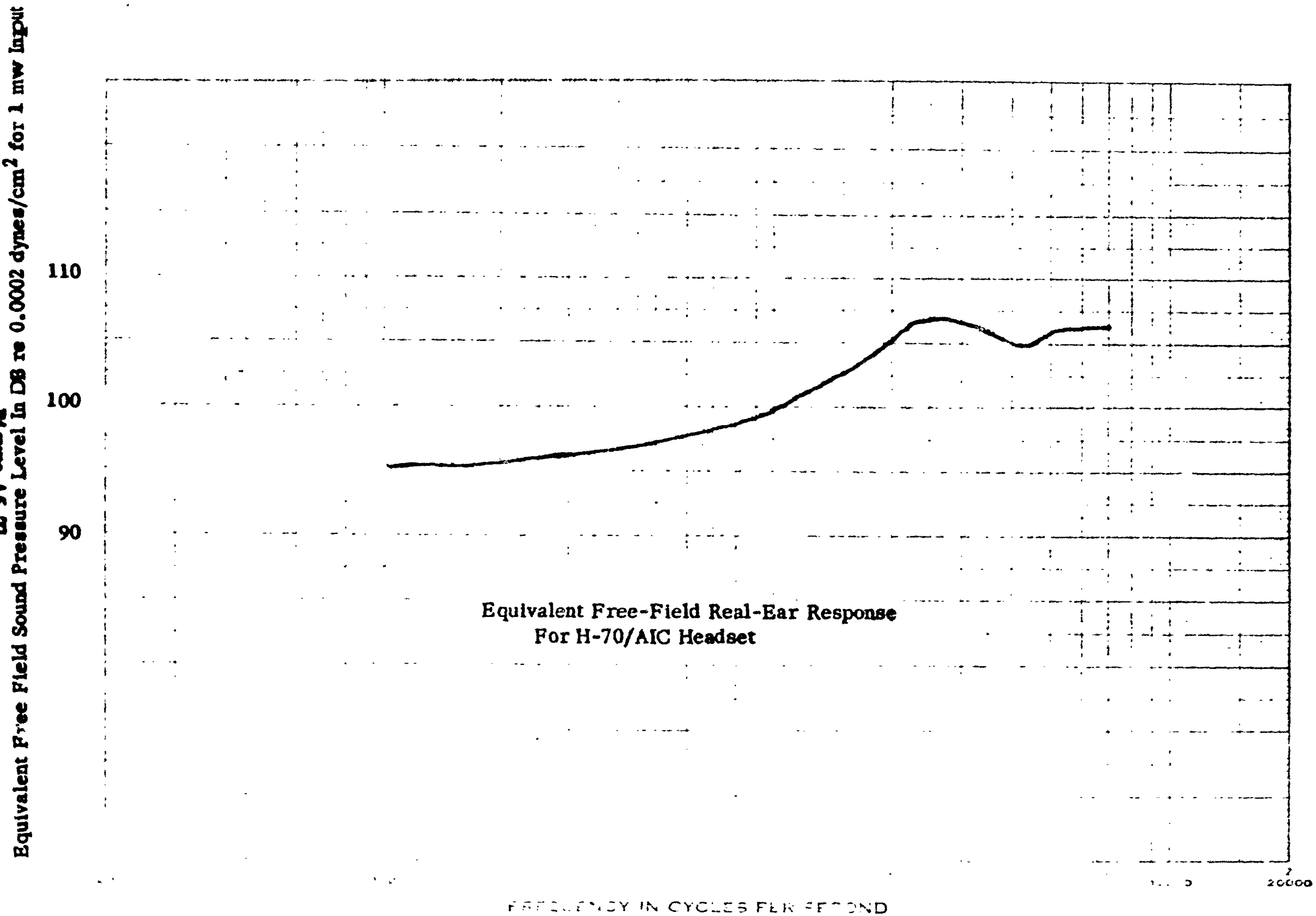

Equivalent Free Field Sound Pressure Level in DB re 0.0002 dynes/cm² for 1 mw Input
110
100
90
Equivalent Free-Field Real-Ear Response
For H-70/AIC Headset
20000
FREQUENCY IN CYCLES PER SECOND

Figure A5-71

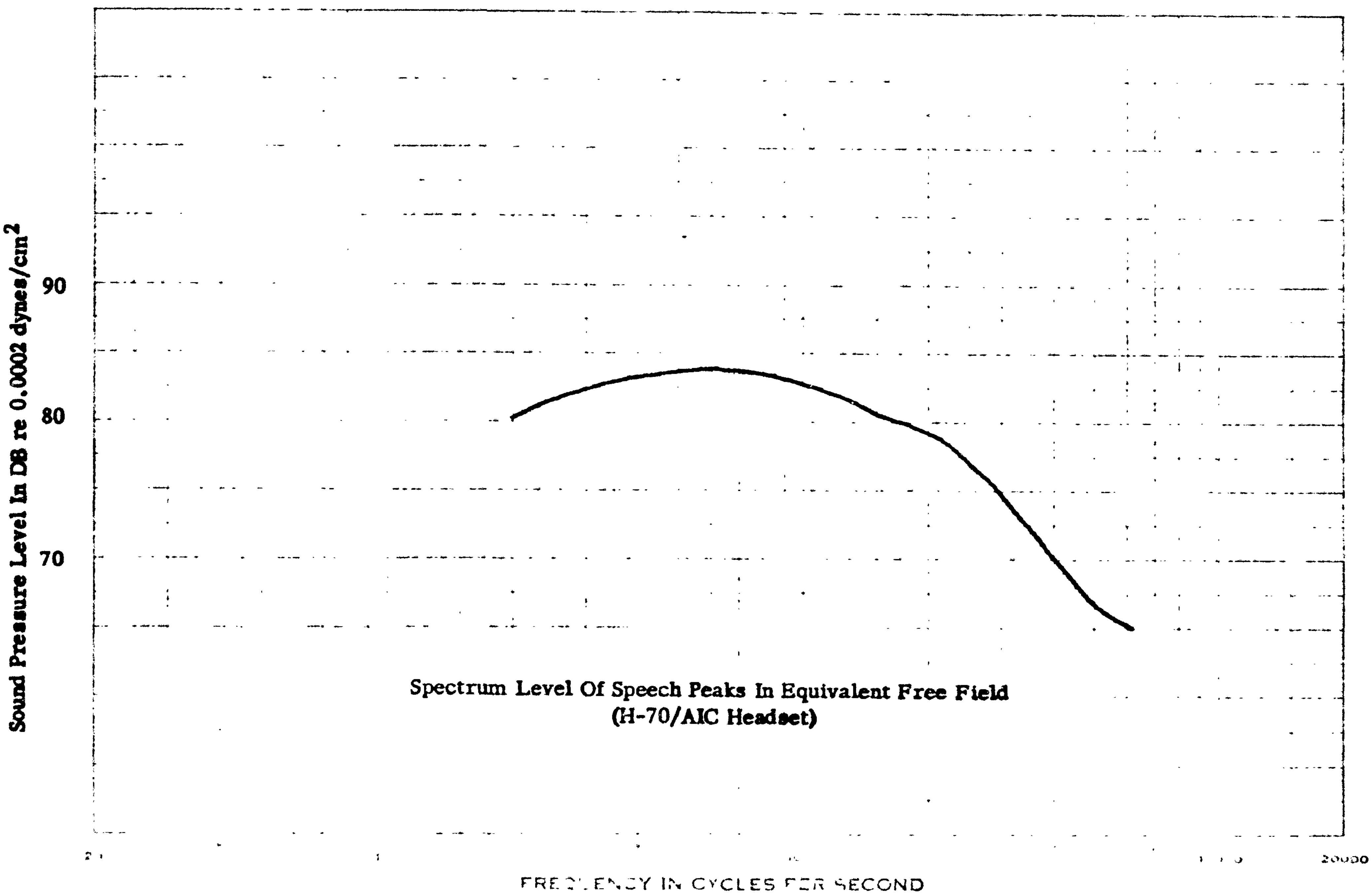
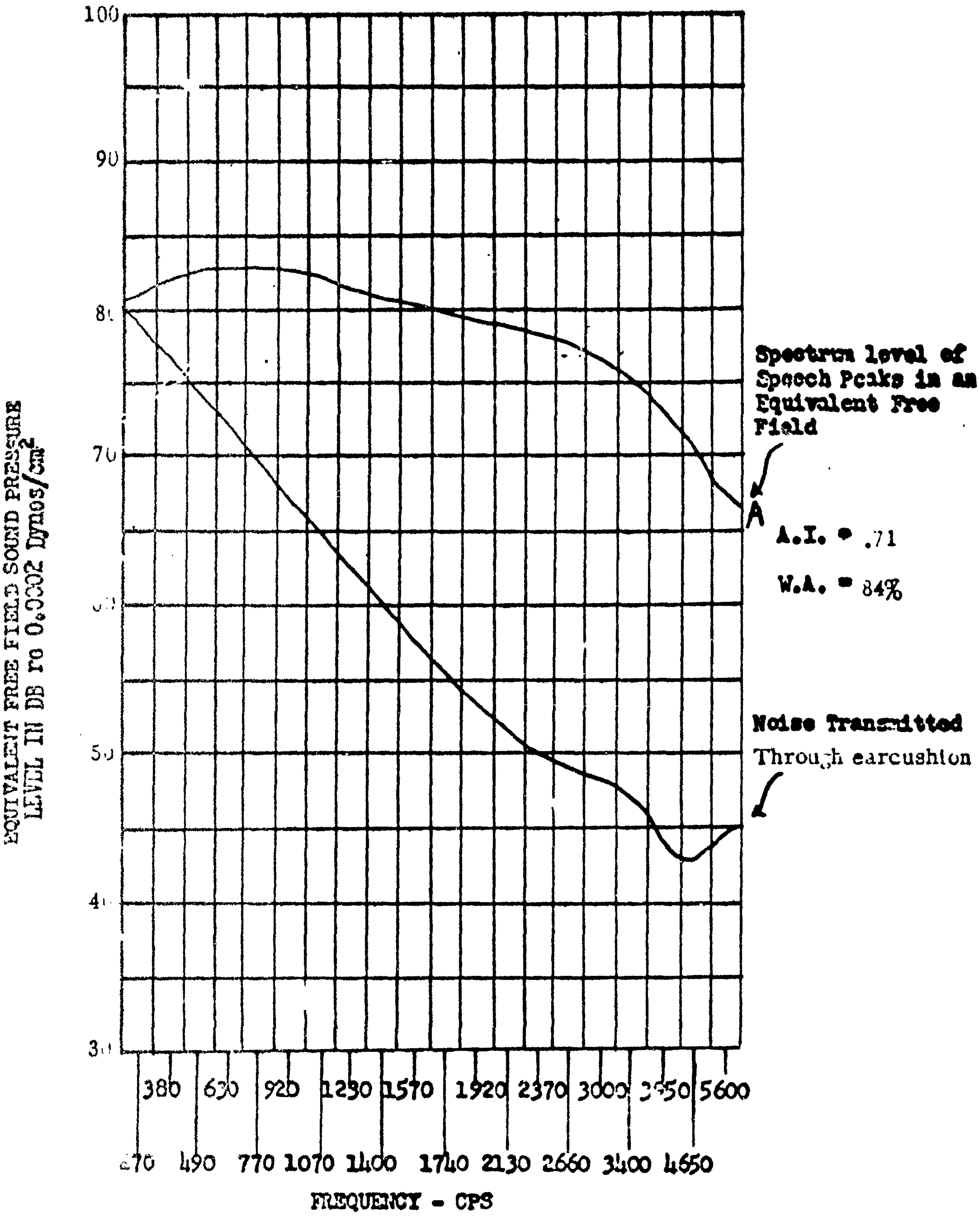

90
80
70
Spectrum Level Of Speech Peaks In Equivalent Free Field
(H-70/AIC Headset)
20000
FREQUENCY IN CYCLES PER SECOND
Sound Pressure Level In DB re 0.0002 dynes/cm^2

Figure A5-72



ARTICULATION INDEX COMPUTATION CHART
FOR
H-70/AIC Headset- 3 Ring Fiberglass Earcushion (200 milliwatts, no clipping)

Figure A5-73

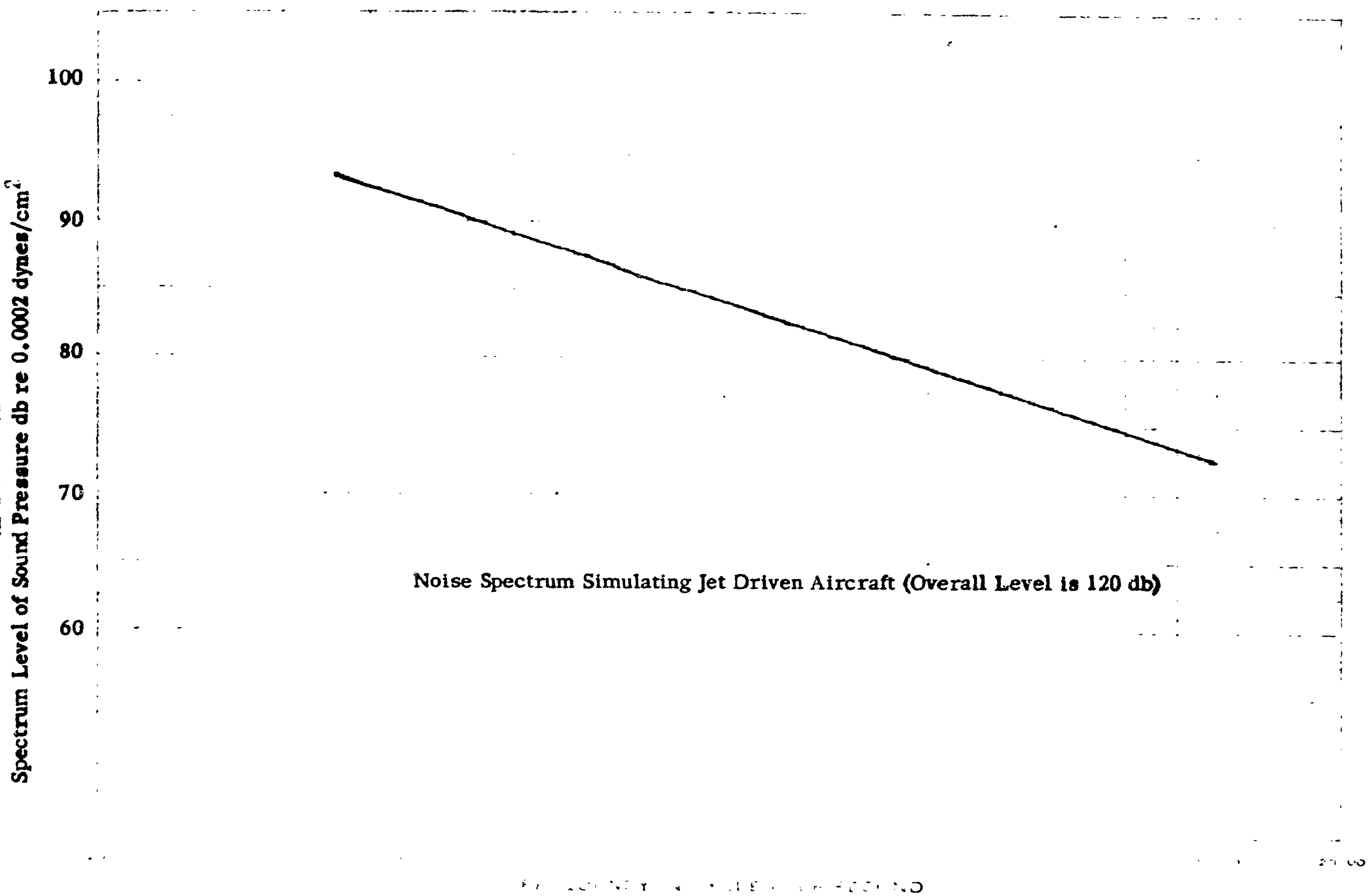

100
90
80
70
60
Spectrum Level of Sound Pressure db re 0.0002 dynes/cm^2
Noise Spectrum Simulating Jet Driven Aircraft (Overall Level is 120 db)

Figure A5-74

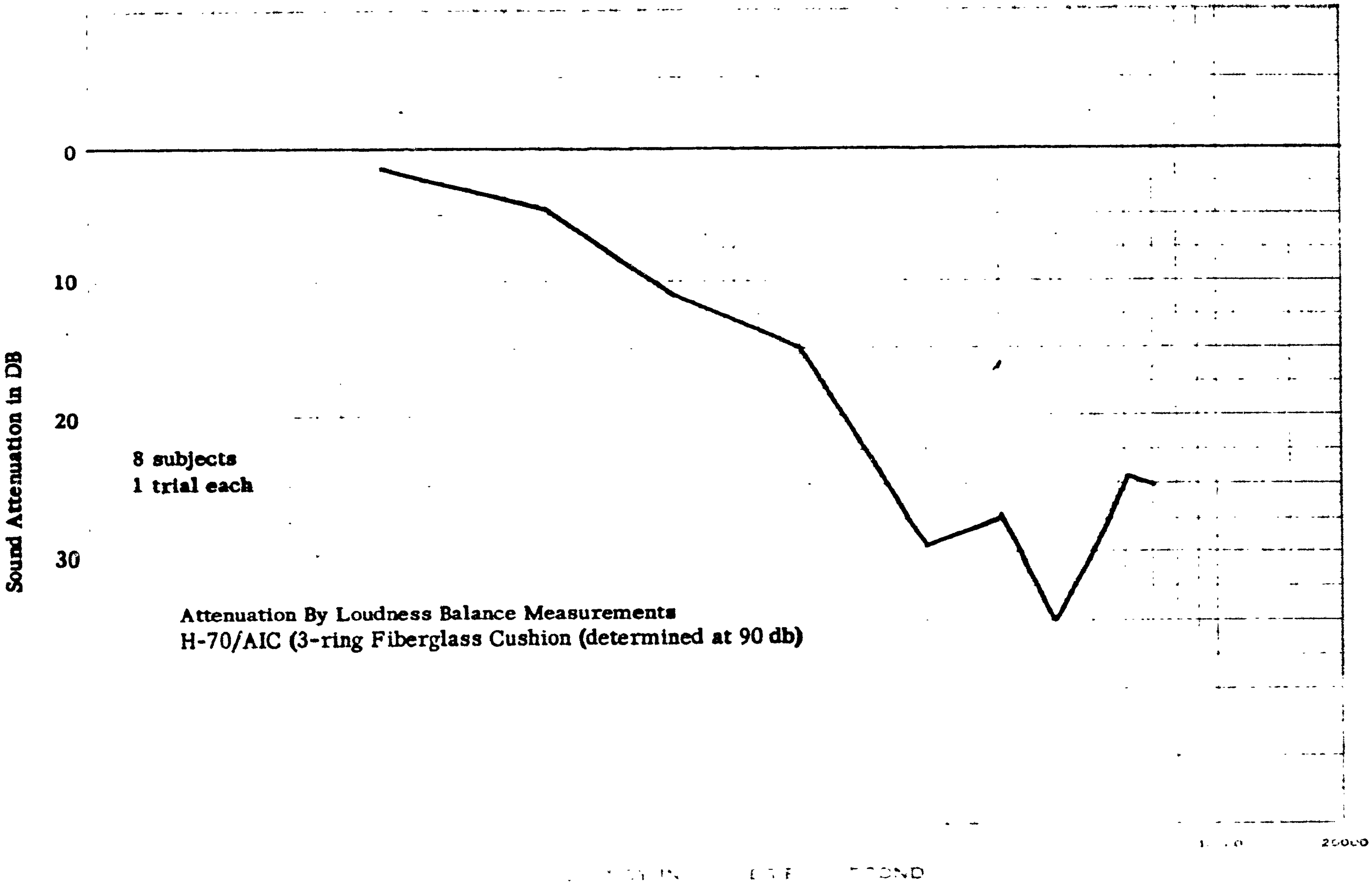
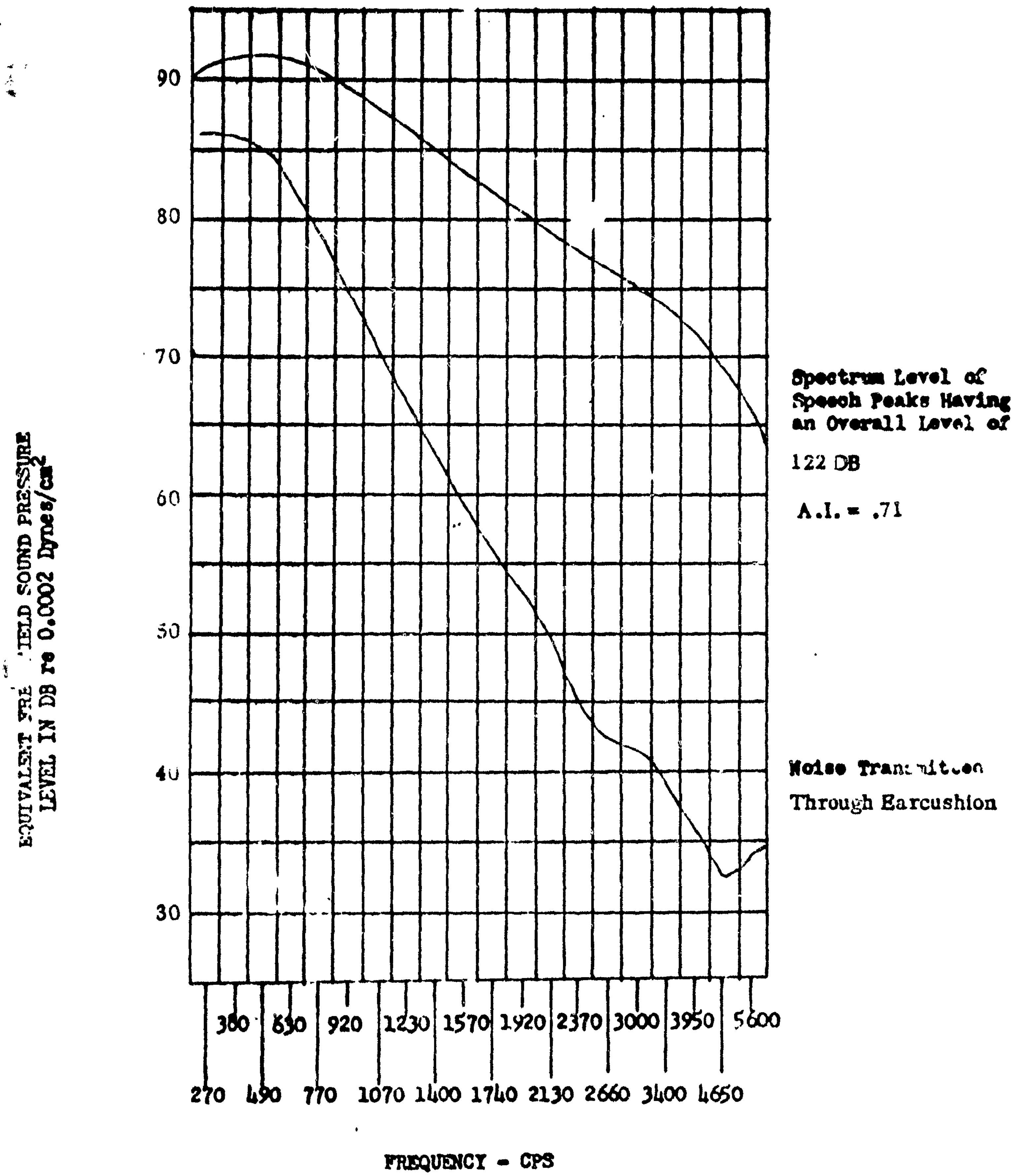

Sound Attenuation in DB
0
10
20
30
8 subjects
1 trial each
Attenuation By Loudness Balance Measurements
H-70/AIC (3-ring Fiberglass Cushion (determined at 90 db)
20000

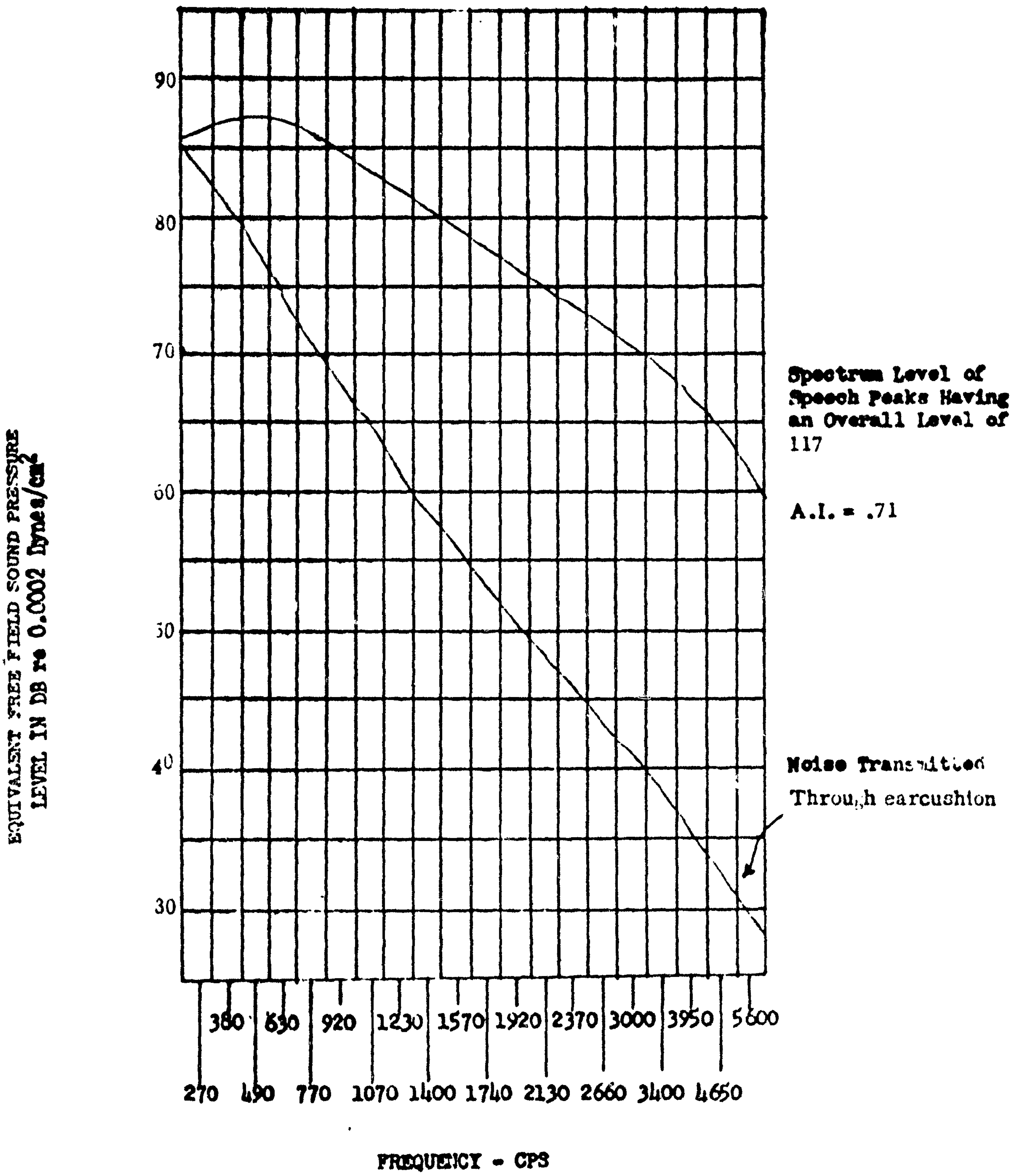
Figure A5-75



ARTICULATION INDEX COMPUTATION CHART
FOR

ANB-H-1 - MC-162A Earcushion

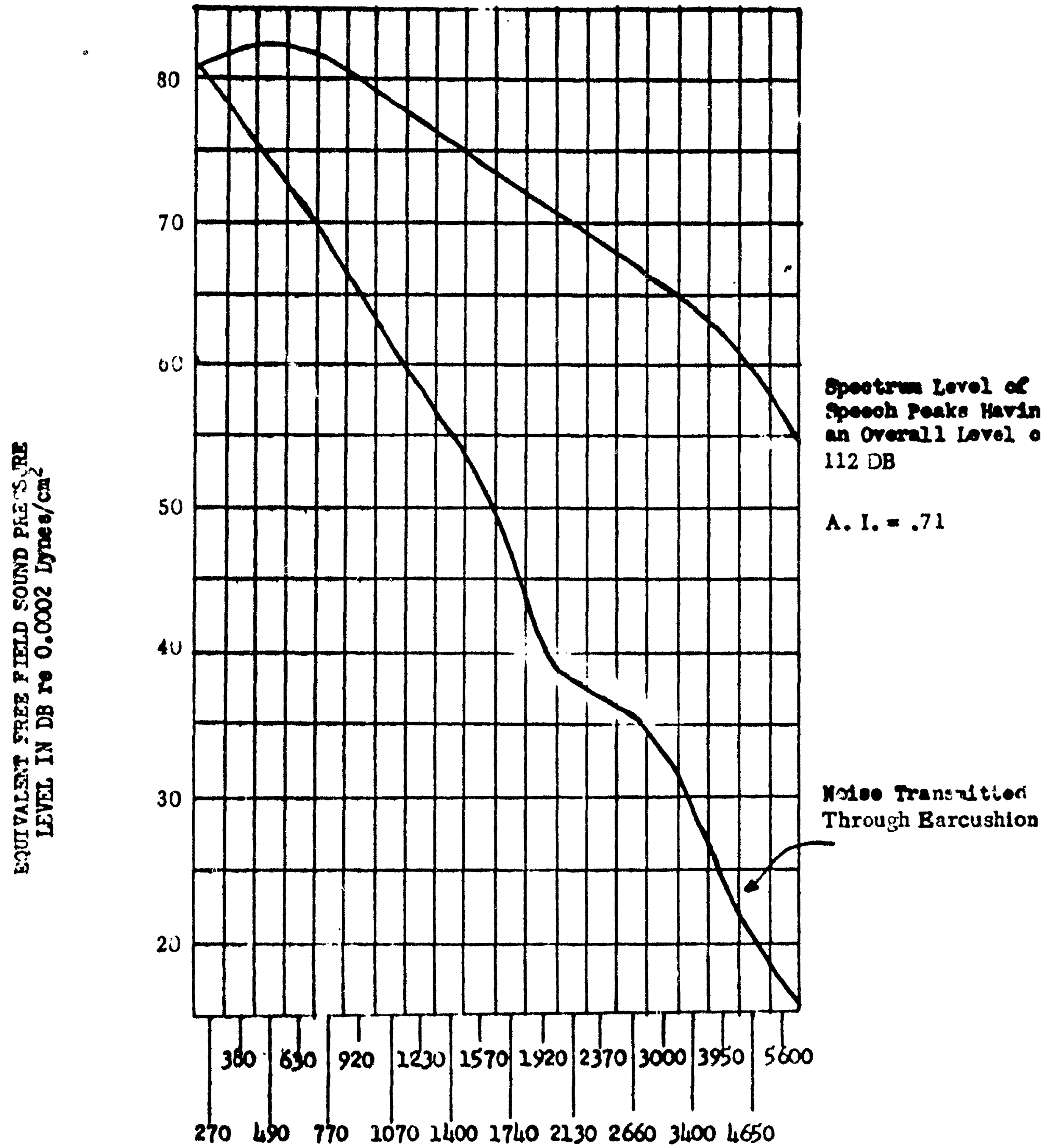
Figure A5-76



ARTICULATION INDEX COMPUTATION CHART FOR

ANB-H-1 - Harvard Design 5-B

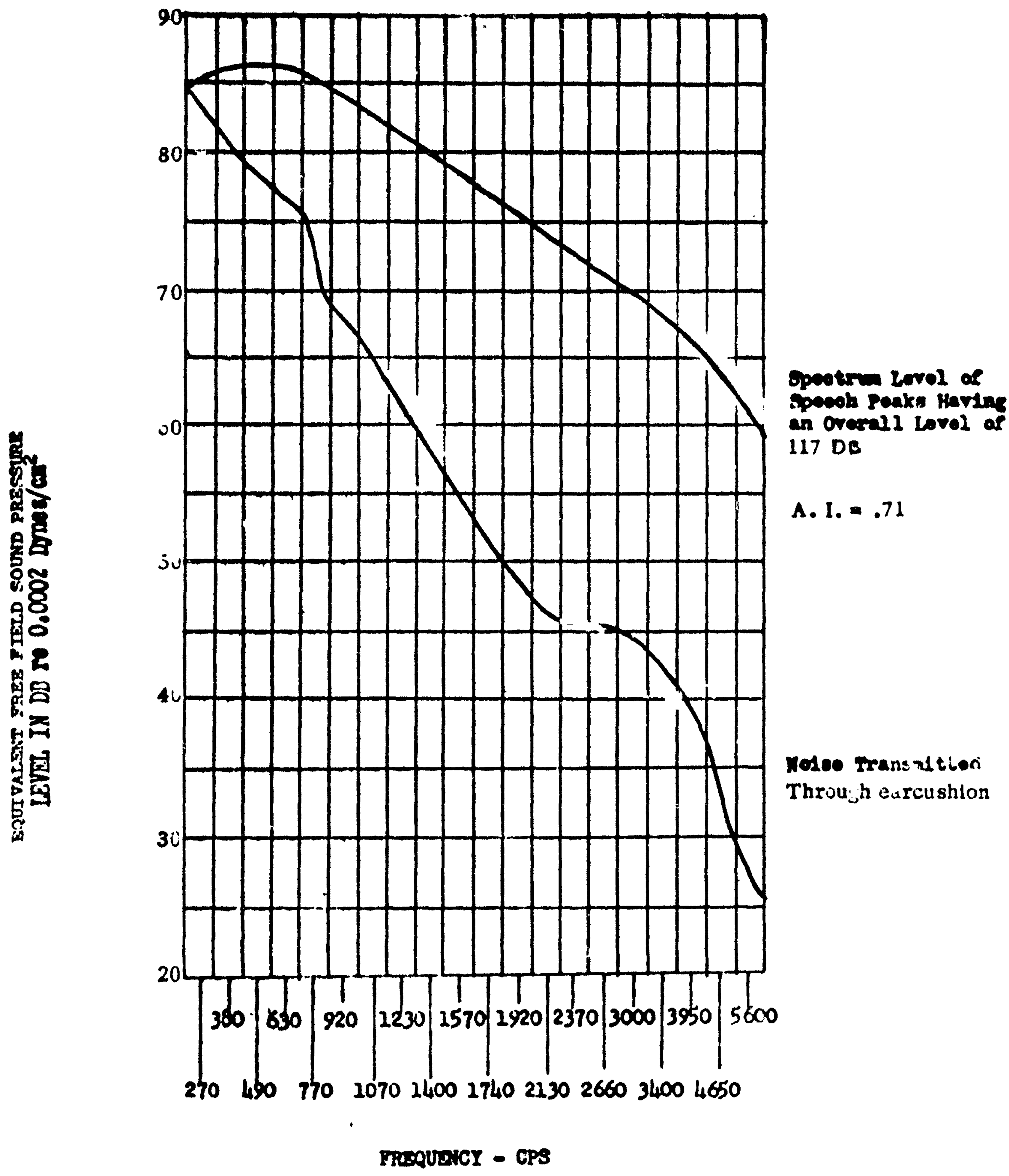
Figure A5-77



ARTICULATION INDEX COMPUTATION CHART
FOR

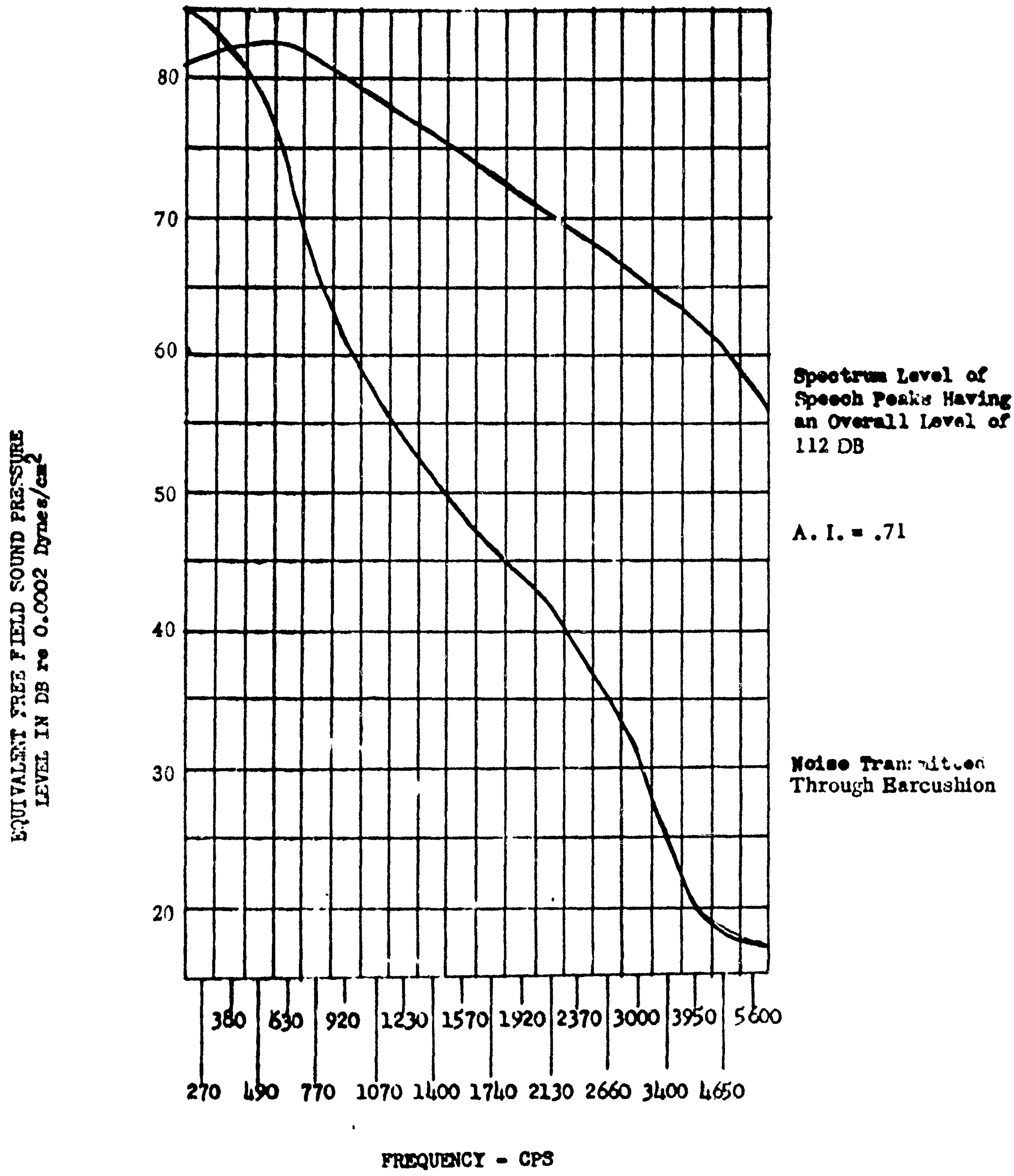
ANB-H-1A - MC-162A Earcushion

Figure A5-78



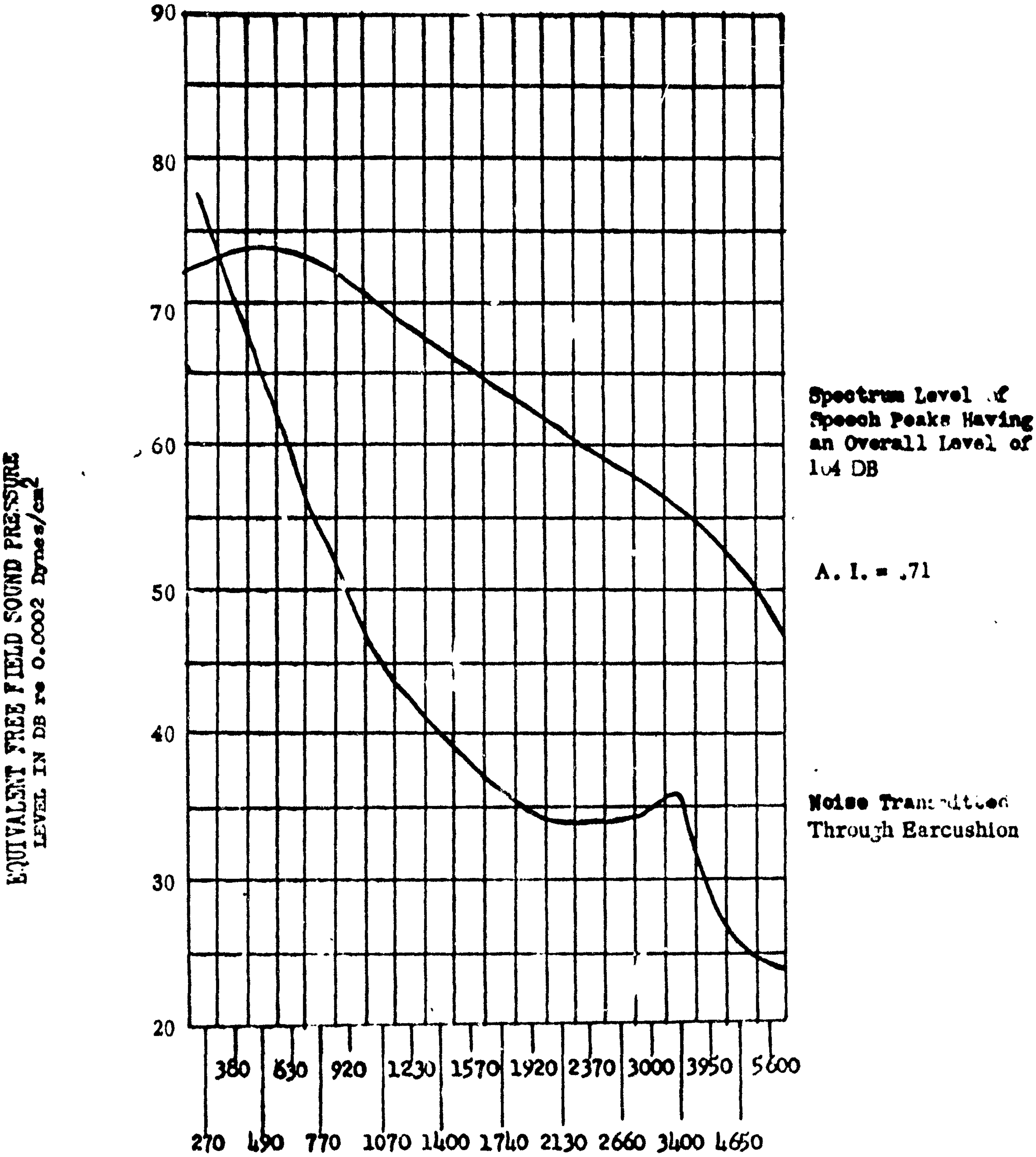
ARTICULATION INDEX COMPUTATION CHART
FOR
ANB-H-1A - Harvard Design 5B

Figure A5-79



ARTICULATION INDEX COMPUTATION CHART
FOR
H-78B/AIC Headset

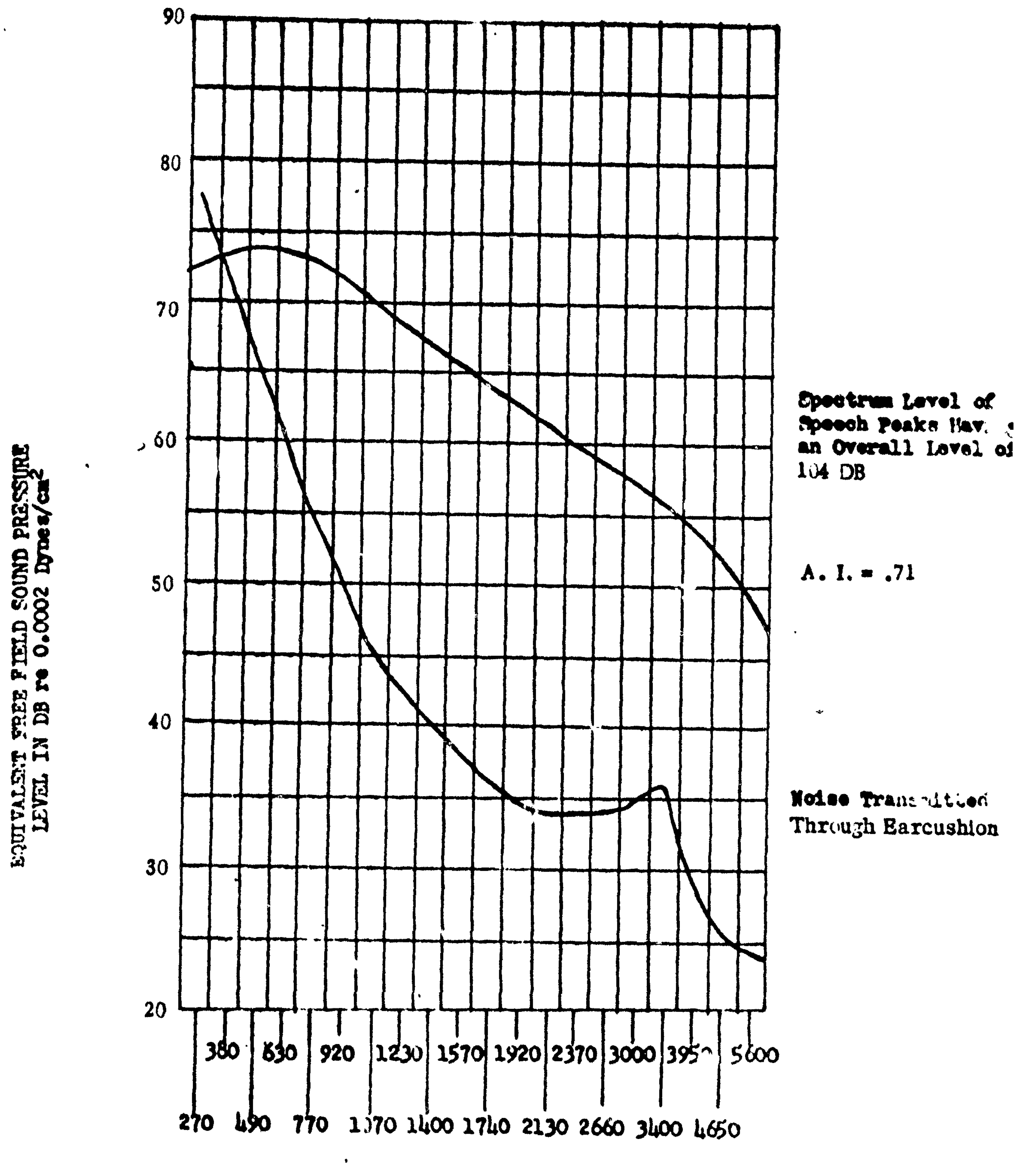
Figure A5-80



ARTICULATION INDEX COMPUTATION CHART
FOR

H-158 Headset - Liquid Pad ...

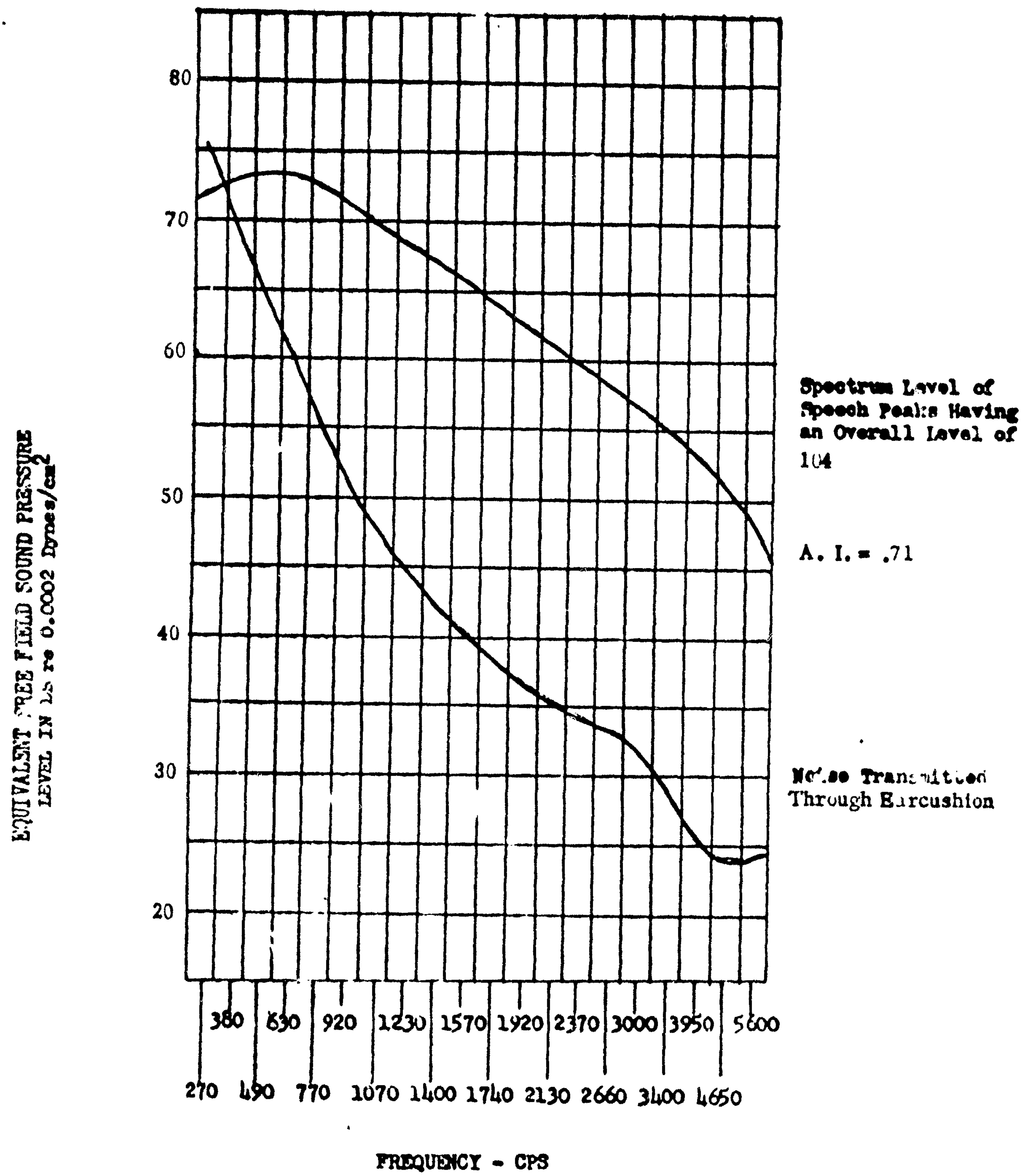
Figure A5-81



ARTICULATION INDEX COMPUTATION CHART
FOR

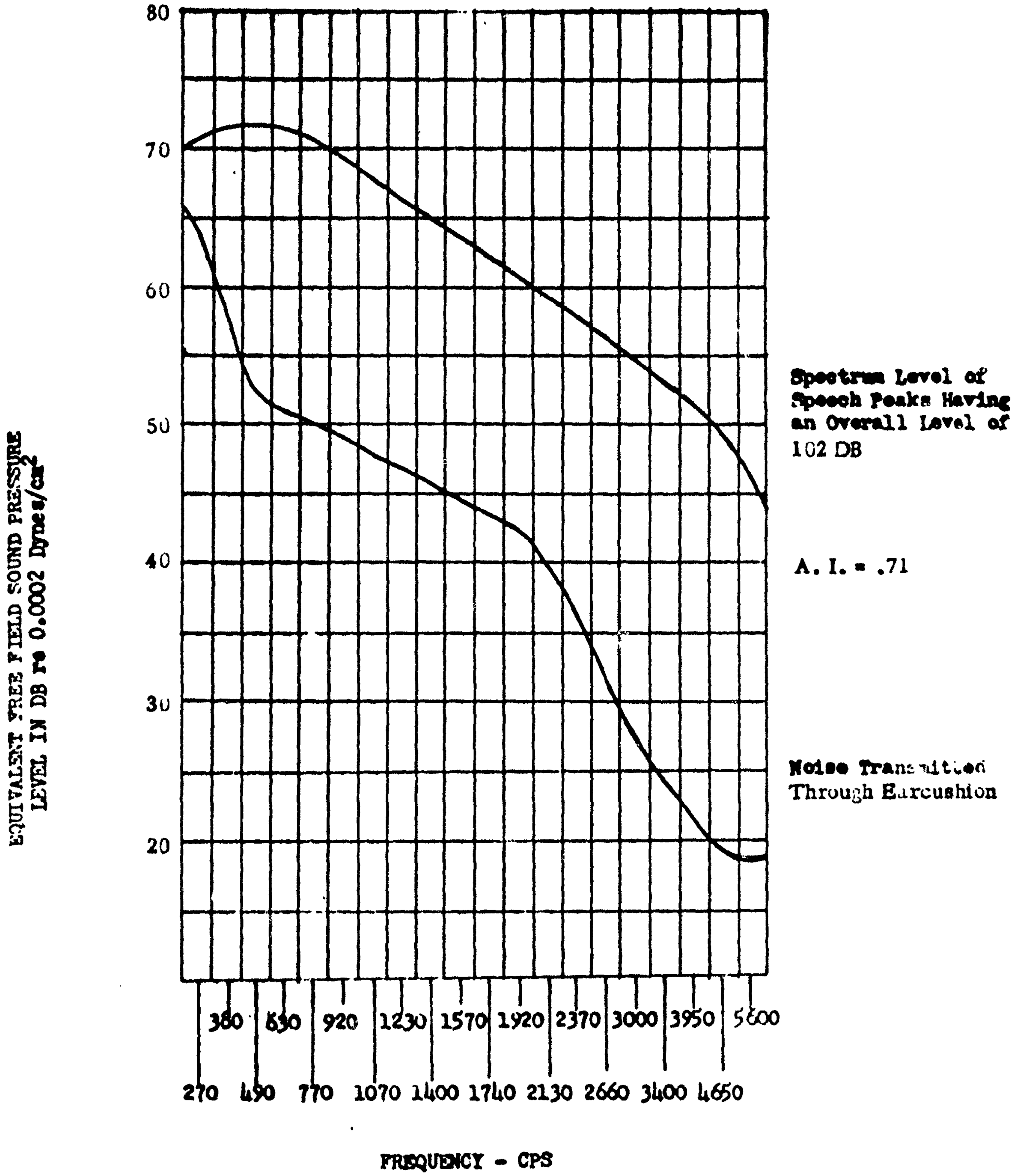
H-158 Headset - Liquid Pad ...

Figure A5-81



ARTICULATION INDEX COMPUTATION CHART
FOR
H-153 Isocyanate Pads

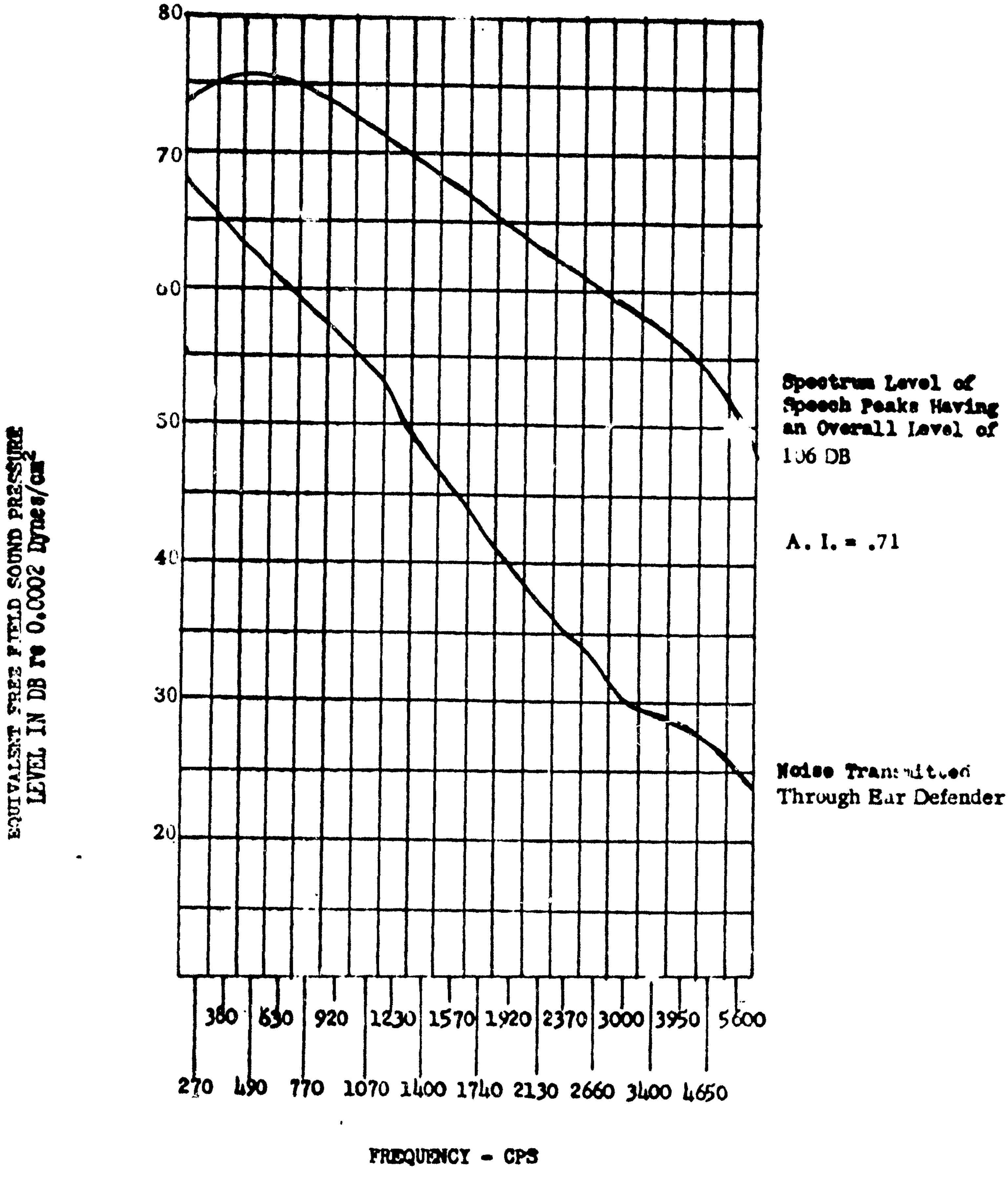
Figure A5-82



ARTICULATION INDEX COMPUTATION CHART FOR

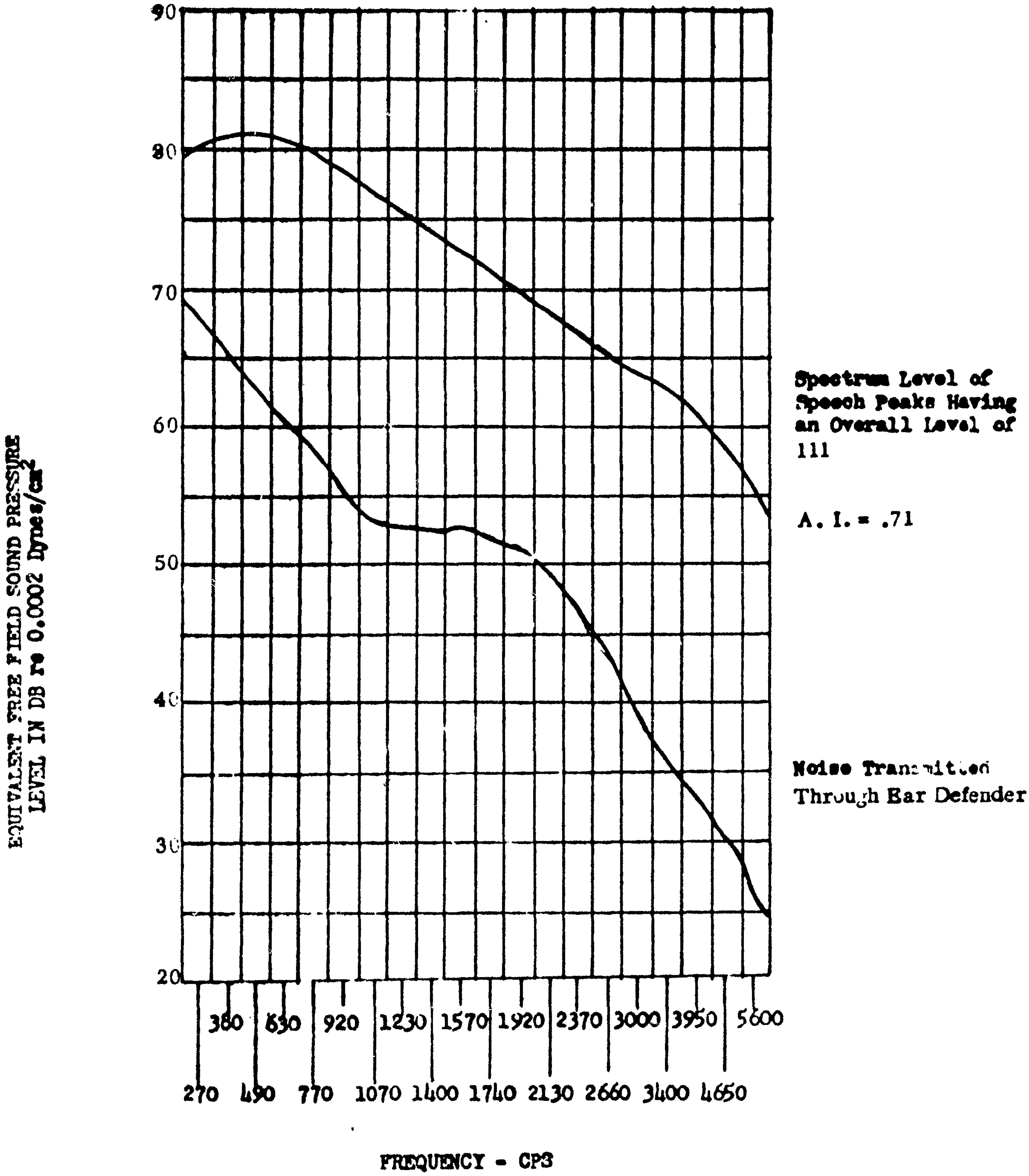
David Clark - Model 372-8C-M

Figure A5-83



ARTICULATION INDEX COMPUTATION CHART
FOR

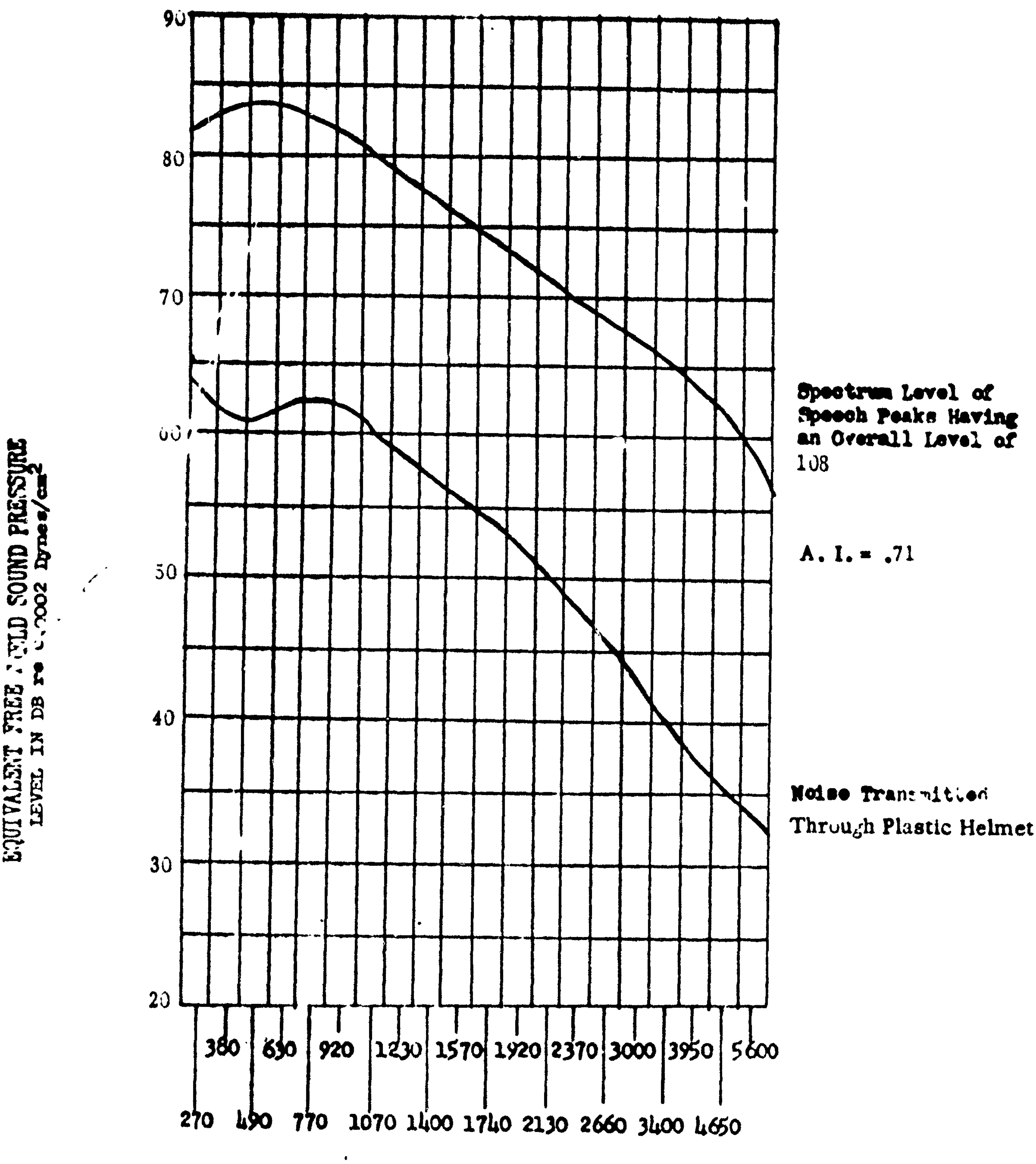
MSA Ear Defender
Figure A5-84



ARTICULATION INDEX COMPUTATION CHART
FOR

Harvintip - M-3 Double Flange

Figure AS-85



ARTICULATION INDEX COMPUTATION CHART FOR

Experimental Plastic Helmet

Figure A5-86

APPENDIX 6.0

STUDY OF HELMET ATTENUATION AND LOUDSPEAKER COMMUNICATION

BY

ARMOUR RESEARCH FOUNDATION

APPENDIX 6

STUDY OF HELMET ATTENUATION AND LOUDSPEAKER COMMUNICATION

Figures

Tables

Appendix

6.0 HELMET ATTENUATION AND LOUDSPEAKER COMMUNICATION BASED ON PROGRESS REPORTS FROM WORK ASSIGNED TO ARMOUR RESEARCH FOUNDATION

6.1 Helmet Attenuation

6.1.1 Introduction

In the full pressure oxygen helmet, the head is completely enclosed by the helmet, and hence, the helmet becomes a major factor in noise exclusion.

In general, the helmet encloses the receiver units in their position on the ears, and the microphone in its position in front of the lips. Therefore, the helmet controls the signal to noise ratio for both the speaking and listening ends of a communication link. Hence the improvement of the attenuation of helmets is a vital aspect of the communication system.

It has been shown that presently available helmets (such as the MA-1 described in appendix 4), represent a very limited acoustical concept and should be subject to considerable improvement in attenuation.

This section reports results of work to determine how the acoustical performance of helmets may be improved and in a general way, the limits which may be expected. Physical tests were conducted to determine the effect of certain helmet parameters on the attenuation. These included size and shape, material, seal around shoulders, and internal absorption.

Several enclosures were tested and an experimental plastic helmet which could be worn by a human subject was designed, made, and tested.

6.1.2 Apparatus and Procedure.

The physical measurements on the helmets described below were made in the reverberation room at Armour Research Foundation. This room is described in the literature. The standard jet noise spectrum was introduced through a horn loudspeaker system.

The attenuation of the helmets with no head present was measured in the following manner. Measurements were made of the sound pressure level inside the helmets with a condenser microphone so placed that its face was at the center of the helmet. The resulting attenuation was abtained for each octave band by comparing this sound pressure level with that existing at the same location in the room without the helmet.

6.1.3 Evaluation of Parameters Influencing Helmet Attenuation

6.1.3.1 Size and Shape

To determine what influence, if any, the shape and size of the helmet might be expected to play in attenuation, two different shapes and sizes of hollow glass globes of the type used to cover street lights were obtained. Globe A was spherical in shape with a diameter of 12 inches, Globe B was a truncated sphere roughly the shape of the ornament often found on top of gasoline pumps. The diameter was 14 inches and the thickness about 4.[11] The opening at the bases of these globes were approximately 4 inches in diameter. They are shown in the photograph of figure A6 - 1. The attenuations of these globes is shown in figure A6 - 2. The spherical shape gave from 4 to 13 db more attenuation in the octave bands above 600 cps. than the truncated sphere.

For the various flexural modes of vibration of a shell the volume changes associated with the vibration are probably least for the spherical shape due to its inherent ridgidity. Hence, one would expect the spheres shape to give improved attenuation over a trundated sphere.

6.1.3.2 Material

For a spherical shell subjected to a uniform hydrostatic pressure, the radial displacement due to the pressure is given by the formula:

$$\text{Radial displacement} = \frac{R^2P(1 - U)}{2\,Et}$$

where R = average radius of sphere
P = pressure
t = thickness of shell
E = modulus of elasticity
U = Poisson's ratio

For a given sized sphere, the radial displacement under a given pressure is inversely proportional to the product of the modulus of elasticity of the material and thickness of the shell. If we wish to decrease the amplitude of response of the shell to applied pressure we must decrease this product. The "stiffer" the shell in this sense the less its change in volume when subjected to pressure and the less the transfer of energy into the interior of the shell. Of course, this concept holds only for the lowest radial mode of vibration.

With this factor in mind two spherical helmets were compared. 1. An experimental plastic helmet 14" in diameter with a thickness of .035". This helmet is discussed in detail in section 6.1.4.

2. A steel helmet 14" in diameter with a thickness of .035".

The attenuation of these two helmets as measured experimentally is shown in figure A6-3. The attenuation of the two helmets was nearly the same for frequencies below 600 cps. The attenuation minimum for the plastic helmet occurs in the 1200-2400 octave band, the minimum for the steel helmet occurs in the 2400-4800 octave band. It would appear that the fundamental radial mode for the steel sphere is higher in frequency by the order of an octave over that of the plastic sphere.

The frequency of the lowest radial mode of a thin spherical shell is given by:

$$f = \sqrt{\frac{E}{2\pi^2 a^2 \rho (1-\sigma)}}$$

E = Young's Modulus
a = radius of sphere
ρ = density of shell
σ = Poisson's ratio

Substituting values for the constants above for steel and plastic indicates that the resonant frequency of the lowest radial mode of the steel sphere is roughly 2.7 times that of the plastic sphere.

6.1.3.3 Seal

It has been shown in a previous section (Appendix 4.2) that the attenuation of a helmet is limited by the nature of the seal at the base of the helmet to the wearer's body. Hence, this problem was explored experimentally.

The seal used with an operational helmet such as an MA-1 consists of a thin rubber collar and a heavier canvas collar. Although this seal is airtight, the attenuation of this seal are poor especially at low and middle frequencies, due to the flimsiness and low density of the material.

Three other types of materials were tested:
1) A seal of foam rubber
2) A captive water seal
3) A captive putty seal

The attenuation obtained on the experimental plastic helmet strapped to the wooden form using these materials is shown in figure A6-4.

The foam rubber seal consisted of a 2" thick pad with a hole in the center to allow entry of the head or microphone.

The water seal consisted of a vinyl plastic doughnut filled with water. The data shown in figure A6-4 is the average of the curves of figure A6-5. In this figure the attenuation of water seal as a function of width/thickness ratio is plotted. From the trend of this data, it would

appear that a rather wide ring will offer superior attenuation with reduced weight.

The third seal consisted of a plastic putty material, sold under the trade name of Mortite.

The results of figure A6-4 indicate that the water seal offers increased attenuation at low frequencies while the foam rubber seal is the better at middle and high frequencies. This effect is due to the increased absorption provided by the foam rubber seal.

6.1.3.4 Absorption

In the data presented thus far, a dip in attenuation always occurs at high frequencies due to the modes of the helmet shell. The introduction of sound absorbing material within the helmet apparently reduces the effect of these modes. The increase in attenuation when 1 square foot of 1" thick fiberglas is placed within a helmet is shown in figure A6-6 for the experimental steel and plastic helmets.

6.1.4 Experimental Plastic Helmet

6.1.4.1 Description

On the basis of some early measurements it was decided that significant attenuation might be achieved by using a spherical helmet. Such a helmet was therefore designed and built. The helmet consists of a transperant plastic sphere 14 inches in diameter attached to a bent piece of plastic which supports the whole helmet on the shoulders. A wooden form was made in the general contours of the shoulders to support the helmet for tests of sound attenuation. The only openings into the helmet are two pipe fittings for the ventilating air and the opening for the head. The latter opening is convenient for inserting a microphone to made direct measurements of the sound attenuation. The helmet is shown in Figure A6-7.

6.1.4.2 Attenuation of the experimental plastic helmets on human subjects

The attenuation of the experimental plastic helmet worn by human subjects was measured by placing a microphone near the ear of an observer. Results are shown in Figure A6-8.

The attenuation of the helmet with foam rubber seal was also evaluated subjectively by the threshold shift method. These results agree reasonably well with the physical measurements made with the microphone.

6.2 Loudspeaker Communication

6.2.1 Apparatus and procedure

6.2.1.1 Synthetic jet noise

Noise spectra of jet aircraft were simulated in a reverberation room. The electrical output of a white noise generator was passed through a filter circuit which shaped the spectrum and into two power amplifiers. The output of one amplifier drove a University multiple horn loudspeaker to provide the high frequency part of the spectrum. The other amplifier fed a special 15 inch low frequency speaker. With this arrangement an overall sound level of over 120 db re .0002 microbar could be produced more or less uniformly throughout the room. This sound, which also existed at the position occupied by an observer, had the octave band spectrum shown in figure 12 . The noise simulating the jet airplane was generated independently from the speech producing system.

6.2.1.2 Speech signals

Speech signals were presented to an observer in the reverberation room through a system completely separate from the above noise system. The speech system comprised a Magnecorder, Model PT6-J, an octave band filter, attenuators, amplifiers, and one of two loud-speakers selected as typical oftheir type. These two speakers were, specifically, an Electro-voice Model T25-A driver with Model 6HD horn, and a Bozak Model B209 mid frequency speaker. Speech signals for the tests were pre-recorded on magnetic tapes.

The first tape was recorded through a flat system giving pure unaltered male speech. To establish the level of the speech recording, a short interval of white noise was first recorded. White noise was selected as the reference because it affords a convenient reproducible reference. The words were all recorded with equal effort, at a level such that average maximum speech signals were approximately the same when monitored by A VU meter as the average of the white noise previously recorded on the tape. No carrier sentence was used with the words. A time interval was provided between words to allow for a response from the listener. The speech tape was played into the room at a level previously determined by the level of the noise recorded on the tape.

This method of measuring the speech levels has since been checked for PB words and the long time average for the "Joe - lawn" sentence. The Joe - lawn sentece was spoken with fixed effort in to a Kellog microphone and the output of the microphone

recorded on magnetic tape. With this same speech effort PB word lists were recorded on the tape using a VU meter to monitor the level. The signal level of this tape was measured with a VU meter having a 1000 mf. capacitor across the meter movement. The indicated VU level for the Joe - lawn sentence was then three db below the VU level of white noise with frequency between 600 and 2400 cps, recorded on the same tape. In addition, the Joe-lawn sentence and the noise as recorded on the tape were run through a square law integrating circuit similar to one described by Benson and Hirsh. A time constant of one second was used. The indicated level for the Joe - lawn sentence was 4.5 db below that of the noise. Other methods might have been used to monitor the level of the speech test material. However, there is a definite relationship between various methods as shown by Benson and Hirsh.

6.2.1.3 Level Measurement

In addition to the two systems used for generation of jet noise and speech signals, respectively, a third circuit was used for monitoring the noise levels in the reverberation room. A Kellog condenser microphone with its associated amplifiers and calibration facilities provided the electrical voltages proportional to sound pressure level which were read on a Ballantine Model 300 voltmeter. Sound pressure, SPL, of the noise, as well as the speech were measured by this system. SPL's were converted into sound power by means of the following formula:

$$10 \log W = L + 10 \log A - 136.5$$

W = acoustic power in watts

L = sound pressure level in a reverberant field in db re .0002 microbar

A = measured absorption of the reverberation room in sabins.

This relationship between the sound pressure level in a reverberant space and the sound power provides a convenient way of determining sound energy. From this, efficiencies of sound reproducing systems can readily be determined.

The signal-to-noise ratio used here has been determined as follows: The sound pressure level for the noise is the reverberant level as measured by the Kellogg condenser microphone in the reverberation room. This is the sound pressure level that would be measured by the microphone at the position occupied by the observer during the testing procedure. The sound pressure level of the signal is determined by the noise on the speech tape. It is measured at the

position to be occupied by the observer's head. The signal to noise ratio as used in Figure A6-13 and elsewhere in this appendix section is the difference in db between the sound pressure levels as obtained above for the noise and the speech. In terms of a long time average level for speech the signal is three db lower than the above.

6.2.2 Measurements of Loud Speaker Characteristics

Characteristics of importance to specification of communication system performance have been measured for the Electrovoice Model T25A horn type speaker and the Bozak Model B209 mid-frequency speaker. These speakers were used to introduce the speech into the noise environment in the testing for intelligibility of speech. The variation of sound pressure level with distance on the axis of the speaker was measured for both speakers in the reverberant room. The sound used was the noise recorded on the speech tapes used in the testing for intelligibility. The polar response of both speakers at various distances and at various frequencies were also measured. This testing was carried out in the anechoic chamber. These results, given in Figures A6-9 through A6-11 need no further elaboration.

In a reverberant sound field in which the sound is truly random the sound pressure will be the same all over the room, except for locations very near the sound source. The field of sound pressure level for the Electrovoice speaker was measured in a plane section parallel to the floor of the room and through a line joining the center of the speaker to the center of the helmet. This field is indicated in Figure A6-12 together with the approximate positions of the plastic helmet in this plane for the three test distances. It is seen that when the distance is 36 inches the helmet is essentially in the reverberant field of the speaker, whereas at six inches the helmet is in a portion of the field of the speaker where the signal level is of the order of four to six db above the reverberant level. Thus, the listener close to the speaker hears direct speech while at greater distances he hears reverberant speech.

Data is given in Table A6-I concerning power input required to establish a sound pressure level of 100 db re .0002 microbar at a specified distance from the speaker for the speakers used in the intelligibility testing. The computation of the values in this table is based on the measured efficiencies of 2.8 % and 0.31 % for the Electrovoice, Model T25A, and the Bozak Model B209, speakers, respectively.

The efficiencies of the speakers were determined by the measurement of the current input to the speaker required to produce a reverberant level of 100 db in the reverberation room used in intelligibility testing. The

electrical input was derived from a Grason Stadler Co. Model 455-13 noise generator filtered through an Allison Laboratories variable HP and LP filter Model 2-BR, and presented to the speaker with filter settings at 75 and 9600 cps, respectively.

The speech tapes used in the intelligibility testing have included a strip of this noise as reference level. It has been determined and previously reported that the long time average of the "Joe.........Lawn" sentence when spoken with the same effort as used with the word lists is 3 db lower than the long time average for the reference strip of noise.

6.2.3 Intelligibility of Speech Using Spondee Words

Intelligibility scores for spondee words were measured using the Electrovoice and Bozak speakers described above. These measurements were made with the listener wearing the experimental plastic helmet. These results and shown in tables A6 - II and A6 -III and figure A6 - 13.

Intelligibility scores for spondee words were also obtained for an observer sitting in a jet noise environment and in the reverberation room but without the protection of the helmet. The results are presented in Table A6 - III. The remarks pertaining to the measurement of signal and noise sound pressure levels for Table A6 - II also apply for Tables A6 - IIIand A6 - IV .

From Table A6 - III it appears that the signal - to - noise ratio required with a masking jet noise to correctly identify 50 percent of a set of spondee words is of the order of -10 to -12 db. This is true irrespective of the levels of the masking jet noise used and the distance between the observer and speaker. It is to be noted that for the largest value of this distance reported in the table, the observer was essentially in the reverberant field of the speaker through which the speech was presented. On the basis of this comparison, we conclude that the S/N ratio required for the correct identification of 50 percent of a set of spondee words is of the order of -11 db and is the same whether the masking noise is white noise or jet noise. This S/N ratio is also independent of the level of the masking noise at least up to the level of 100 db re 10002 microbar. When using the helmet the level of the noise within the helmet is of the order of 100 db when the level outside is 120 db.

Tables A6 - IIand A6 - IIIpresent the results of similar testing but with the observer wearing the experimental helmet. In obtaining the data for Table A6 -III, Model B209 speaker was used. Data is missing for those cases where the signal required to attain the S/N necessary for intelligibility was above the capabilities of the speaker. Results for both speakers lead to the same conclusions, except for the close position of the helmet to the Bozak speaker.

An examination of the data for both speakers reveals that the presence of the helmet seems to have an adverse effect on the intelligibility of spondee words. At levels of 100 and 110 db for the masking jet noise and using the Electrovoice speaker to present speech, and with the observer wearing the helmet, a S/N of -8 to -10 db was required for a score of 50 percent at distances of 13 and 25 inches from the speaker. At these distances, the observer was within the direct field of the speaker. This is to be compared with the requirement of a S/N of -11 db for comparable distances without the helmet corresponding to these two exterior levels would be 80 and 90 db re .0002 microbar, respectively.

The presence of the helmet has affected intelligibility sufficiently so that the S/N required for 50 percent intelligibility is even more evident when the position of the observer's head is 43 inches from the speaker. At this distance, the observer is in the reverberant field of the projected speech. Here, the order of 6 to 8 db more signal was required with the helmet than without for an observer at the same point in the field.

6.2.4 Intelligibility of Speech using PB Words

Tape recordings with lists of PBwords were used as the test material. The level of the recording was set relative to a strip of white noise recorded at a fixed level. The "Joe........lawn" sentence was then recorded with a fixed effort by a talker. Peaks of speech were monitored by a VU meter to be on the average level as the white noise. The words of the lists were then recorded with this same effort. A 1000μf capacitance was placed across the meter movement of a VU meter and the meter readings were used to relate the levels of white noise and the "Joe......lawn" sentence. The average level of the "Joe........lawn" sentence averaged 3 db lower than that of the recorded white noise.

Preliminary testing indicated the possibility of a measurable difference in intelligibility for PB words when heard in the reverberation room and the anechoic room. Furthermore, there is an abundance of data in the literature for intelligibility of PB words with a masking of white noise. In order that our results might be correlated with those in the literature, an intelligibility test was first conducted in the anechoic room using white noise for masking. The spectrum of white noise used is presented in Fig. A6 - 14. The intelligibility test was then carried out in the anechoic room using the synthetic jet noise for masking. In both situations the overall noise level for the masking noise was 90 db. A comparison of the results of these two tests should indicate the effect on intelligibility of shaping the noise spectrum. Following this, intelligibility tests were conducted in the reverberation room using synthetic jet noise for masking. The noise level used was 90 db. The observer listened without the protection of the helmet in all three conditions. The result of this last test was expected to reveal what differences, if any , could be

expected between intelligibility in the two extremes of an anechoic and a highly reverberant environment. The actual environment within an airplane must lie somewhere between these extremes. Finally, the tests were run in the reverberation room with the observer wearing the experimental helmet. In this final category of tests, synthetic jet noise levels of 90 db and 105 db were used. The results here should reveal what changes, if any, in intelligibility could be attributed to the spectral attenuation of the helmet, together with the character of the hearing environment within the helmet itself. If no appreciable modification is present, one then needs to concern himself only with the variation of intelligibility with noise level. In the testing with the helmet the maximum noise level was 105 re .0002 microbar. All listening tests in this series were made with the listener at the same distance from the speaker. A plan of the anechoic room indicating positions of speakers and listeners is included as Fig. A6 - 15. Fig. A6 - 16 gives block diagrams for all circuits used in presenting noise and speech to the listener, as well as that used to measure sound pressure levels. All levels for masking noise and the levels for speech were measured at the position to be occupied by the listener's head in the testing. The latter levels are those produced by the noise recorded on the speech tape,. Four members of the staff of the section were chosen for the listening tests, No attempt has been made to test their hearing by audiometers but three were young adults in their early twenties and the fourth a male adult of 38.

The results of these measurements are tabulated in Table A6 - IV.

For low S/N and for listening in an environment representative of the anechoic room there seems to be a slightly better intelligibility for PB words in jet noise. The advantage disappears for S/N of the order of 0 db. Although the data so far obtained indicates an advantage for white noise at high S/N ratio this is less certain and needs further investigation. The articulation scores for two of the observers, VR and CC, were unusually low at a S/N above +2 db when the masking was by white noise. Their low scores have greatly depressed the averages plotted in Fig. A6 - 17. One of these observers, VR, was tested at a lower level of the masking white noise, 80 db, and had articulation scores at this level equal to those of the other observers, DL and LS, at the higher level of 90 db. It is felt, therefore, that the depressed character of curve for S/N above +2 db may not be correct. The interest in the use of white noise as the masking noise lies in the fact that the literature contains considerable data with relation to its use. It is intended that this phase of the experimentation should enable us to relate the work with jet noise to this information. The article by Hirsh , et al, gives the value of -5 db for the S/N corresponding to an articulation noise and PB words were presented to the listener by earphones. The corresponding value obtained by us

with the listener in an anechoic room was from -2 to -3 db.

This difference is easily explained in terms of the directivity of both noise and speech sources in our experiment within the anechoic room. In an earlier article, Hirsh has shown that the intelligibility of spondee words in a masking white noise depends upon the directions from which the noise and speech come. His experiments were also conducted in an anechoic room, although his noise spectrum was flat to 7000 cps, whereas ours increased 3 db per octave. His data indicates that the threshold of intelligibility for spondee words masked by the noise spectrum chosen varied as much as 9 db as the directions of presentation of noise and speech were varied. This threshold for the partidular manner of presentation chosen in our case, that is, with speech coming from behind and the noise from in front, was second highest among the situations reported by Hirsh. This level was of the order of 3 db above the average for all cases.

This data indicates that our value of -2 to -3 db S/N for an articulation score of 50 percent is high because of the way in which noise and speech were presented to the observer.

For the second step in the testing program, articulation scores were obtained for the four listeners in the reverberation room with a masking jet noise. In this environment the results for all four observers were more nearly alike. The average articulation score against S/N for this case is also plotted in Fig. A6 - 17. The results from all four observers have been used to determine these averages.

The striking feature of jet noise masking in the anechoic and reverberation rooms is the small differences in intelligibility for such widely different environments. The intelligibility appears to be slightly higher for listening in a reverberation room in the range of low S/N. For an intelligibility score of 50 percent the improvement is of the order of 2 to 3 db. Any actual listening environment for personnel operating jet aircraft must, of necessity, lie between these extremes. The differences between extremes is so small that we conclude that the use of the reverberation room for the testing is satisfactory.. Results obtained in an environment such as the reverberation room would not be expected to differ markedly from those to be anticipated in actual use.

In light of the results in the paper by Hirsh, mentioned above, we can conclude that the results when listening in an anechoic environment are even more like those when listening in the reverberation room than curves B and C of Fig A6 - 17 indicate.

Finally, testing on intelligibility using PB words was conducted in the reverberation room with the observer wearing the helmet. Results at two levels of the masking jet noise, 90 and 105 db, have been obtained using two of the four observers. The results when the masking jet noise was at a level of 90 db re .0002 microbar have been plotted in Fig. A6 - 17. Averages of the percent of correctly identified words obtained by the two listeners have been plotted against S/N. A tentative conclusion to be drawn from a comparison of the two curves for jet noise in a reverberation room, one with and one without the helmet, is that, at least for low S/N, the helmet adversely affects intelligibility of PB words. The use of the helmet has resulted in the requirement of additional S/N of about 3 to 4 db in order that the observer may obtain an articulation score of 50 percent. Results of individual observers also show this adverse effect due to the helmet, 3 db in the case of VR and 1 db for CC. At high levels of S/N good intelligibility is obtained for either case.

For the range of conditions considered in these observations the range of average S/N for 50 percent intelligibility was a little more than 3 db. This is not considered to be a particularly significant difference in view of the application for which the study is intended.

6,2.5 Communication Via Air Supply Tubing

Previous experiments had indicated that high signal levels were required to produce adequate intelligibility for speech when this speech is projected by a loudspeaker exterior to the helmet. Any attempt to operate such equipment at high altitudes and at even higher jet noise levels on the ground than are experienced at present would require large increases in efficiency and in power handling abilities of the speaker, as well as added power capabilities in amplifying equipment. An alternative to this seemed to be the introduction of the helmet from the exterior speaker by way of a tube. In the equipment which is the goal of the work this might be the tubes supplying air or oxygen to the wearer of the helmet. Assuming reasonable transmission characteristics for speech by the tube we then need to provide the required signal - to - noise ratio relative to the much lower sound pressure level of the jet noise found within the helmet. For a preliminary investigation of this method of communication, heavy plastic tubing with an inner diameter of one inch, outer diameter of 1 - 1/4 inch, and of various lengths was procured. The driver unit of the Electrovoice speaker Model T25A was attached to one end of this tubing. The other end was coupled to the helmet by way of small rubber tubing. This was only a preliminary expedient for the purposes of testing for feasibility. By way of a T junction air was also furnished through the plastic tube to the listener when he wore the helmet. A schematic diagram of this apparatus is presented as Fig. A6 - 18. The observer was placed in the reverberation room in a jet noise environment of a specified sound pressure level and speech was introduced

to him through the tube. The voltage applied to the driver unit was measured and the percent of words correctly identified by the observer was noted. This procedure was repeated at several levels of the jet noise and for various lengths of plastic tubing.

The results of these experiments have been summarized in Table 46 - VI. The values used in part (a) of Table A6 - VI for the signal - to ½ noise ratio required for an intelligibility of 50 percent have been obtained from Table II. They are averages of the results for the two ob servers. It is evident from Table A6 - V that the same level of intelligibility, an articulation score of 50 percent for spondee words, is attainable by a considerable reduction in electrical power requirements of the speaker if the tube method is used.

Assuming the resistive component of the speaker impedance to be the same for the two condidions of Table A6 - V, the power reduction is between 20 and 35 db. This is the same magnitude as the noise attenuation of the helmet. Thus, the same S/N is obtained but because the noise is attenuated by between 20 to 35 db, the signal may be reduced correspondingly. At 120 db noise sound pressure level in the room, 50 percent of the spondee words were correctly identified with only one volt of speech signal on the speaker feading through the tube. In contrast, 10.7 volts were required for the same intelligibility by a listener 13 inches from the speaker.

6.2.6 Summary of Loudspeaker Power requirements

From the material of section 6.2.2 and the results of the articulation testing, the power required to achieve a given articulation percentage may be determined. Tables A6 - VII and VIII summarize the results.

6.2.7 Intelligibility of Clipped Speech Projected by a loudspeaker in Noise (conducted by WEAL)

In order to utilize the available power in a given transmission system, clipping of speech has been widely used without significant decrease in intelligibility.

Infinite clipping results in a ratio of 1 between peak and average energy. Normally, the clipped speech is fed through a high-pass filter in order to protect the loudspeaker system and a low-pass filter is also used to limit the higher harmonics. Both filterings normally result in slightly increased intelligibility.

6.2.7.1 Instrumentation

The instrumentation used is shown in Figure A6 - 19. Pre-recorded CVC words were differentiated as to prevent the vowels from dominating the clipping. After differentiation the CVC words were clipped and the clipping level was controlled by use of a scope and an attenuator. After clipping the signal was fed through a band pass filter to the amplifier. Output signal fron the amplifier was finally fed to a "University" 15 watt loudspeaker, Type MM-2.

The level of the average voltage fed to the loudspeaker was measured with a WEAL voltmeter (measures average but reads RMS for sine waves, therefore, 1 db was subtracted from the readings).

The level of the peak voltage fed to the loudspeaker was measured with a Heathkit voltmeter (measures peak voltage but reads RMS for sine waves, therefore 3 db was added to the readings).

6.2.7.2 Procedure

Approximately ten lists of 50 CVC words were tape recorded in the quiet, using a 640AA microphone, 1 foot from the lips. Three talkers, MG, TW, WO, were used.

These tapes were then played back through the apparatus shown in Fig. A6 - 19

The clipping level was set in the following manner:

(1) A tape with CVC words was played back and the clipping level control and the gain of the amplifier before the clipper, was adjusted so the very highest peaks in the words were barely clipped. The gain of the amplifier was increased 30 db so the speech was clipped. (This clipping level corresponds to approximately 25 db of clipping where zero db of clipping corresponds to just barely clipping of the average peak level in words.)

(2) The clipped wave was then fed through the filter, and the peak level measured.

(3) This peak level was used as reference voltage for feeding a sine wave to the output amplifier. The sine wave was adjusted to the same peak level as the clipped and filtered speech wave. The gain control of the amplifier was then adjusted so the amplifier delivered 15 watts of continuous energy to the loudspeaker.

(4) The input to the output amplifier was then switched from the sine wave to the clipped, filtered speech wave.

6.2.7.3 Results

The loudspeaker was mounted in the noise enclosure one foot forward of the forehead. The score for a listener (JPC) with a MA Helmet, was 88% under quiet listening conditions. The power for the clipped, filtered speech wave was:

Peak Power:	Approximately 15 watts
Average Power:	Approximately 2.5 watts

$$\frac{\text{Peak Power}}{\text{Average Power}} = 8 \text{ db}$$

The score decreased to 65% when standard jet noise spectrum was introduced. The signal level, noise

level and signal - to - noise level at the helmet are shown in Figures A6 - 21 and A6 - 22.

References for Appendix 6

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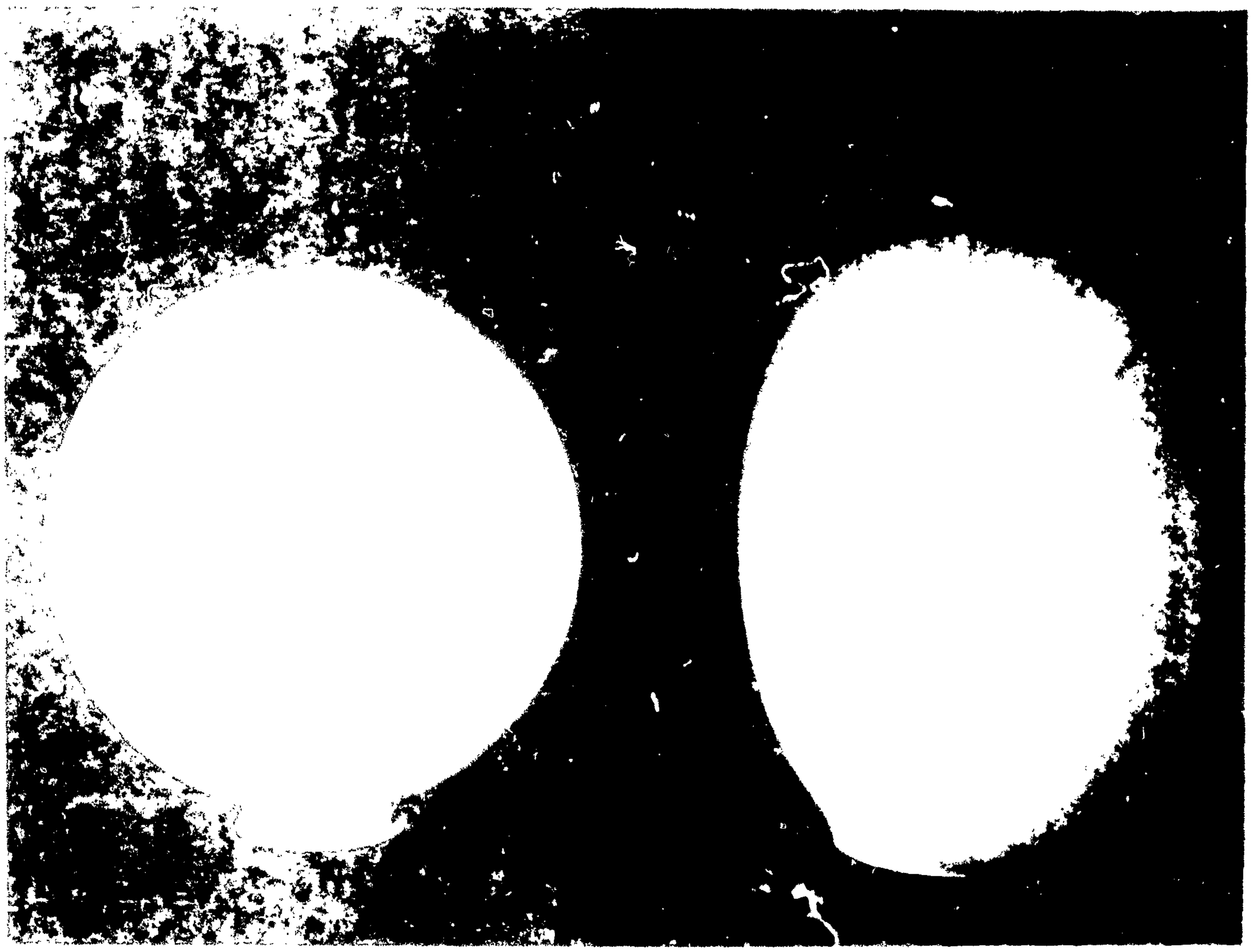
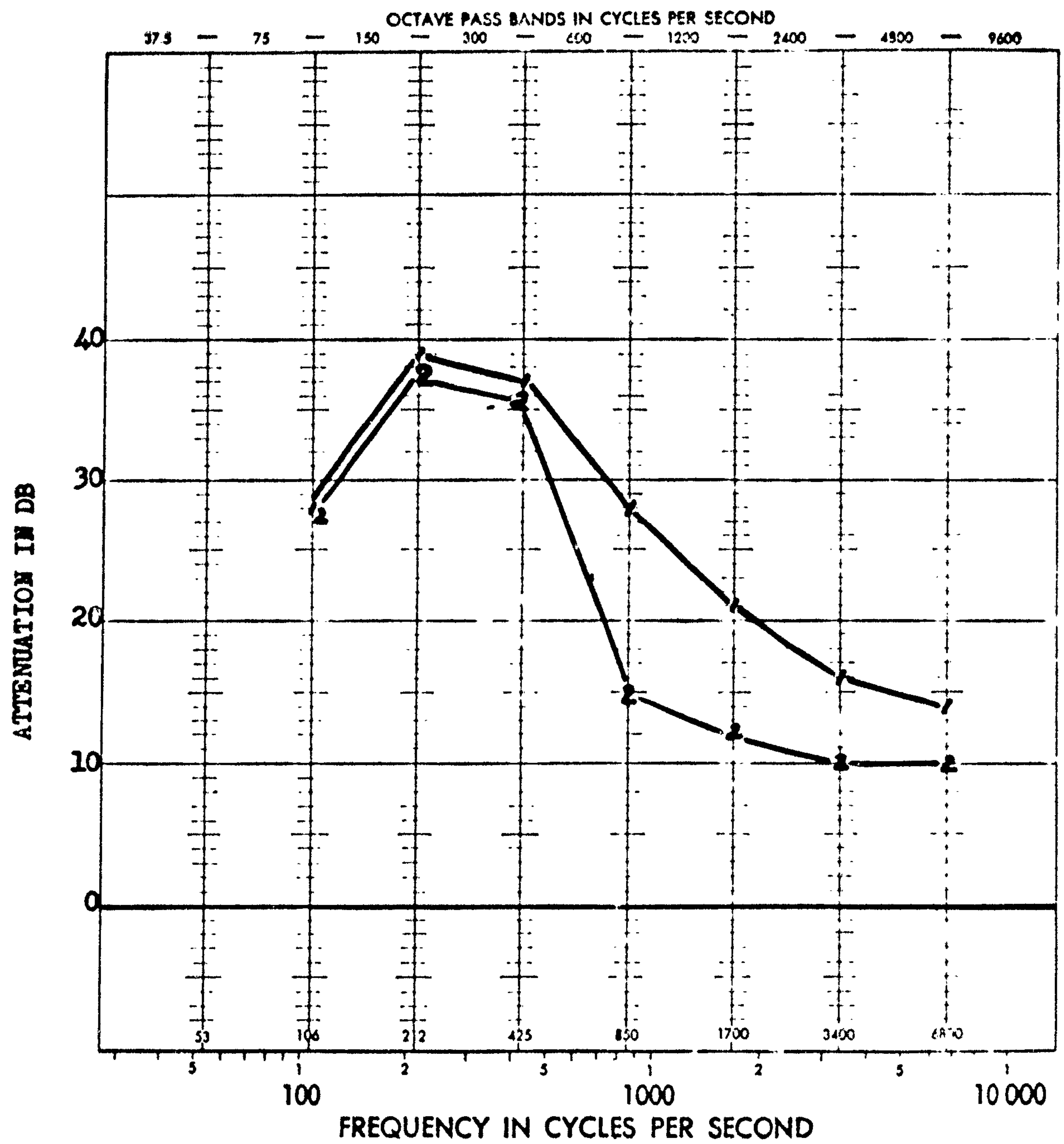


ATTENUATION OF NOISE BY EXPERIMENTAL HELMETS

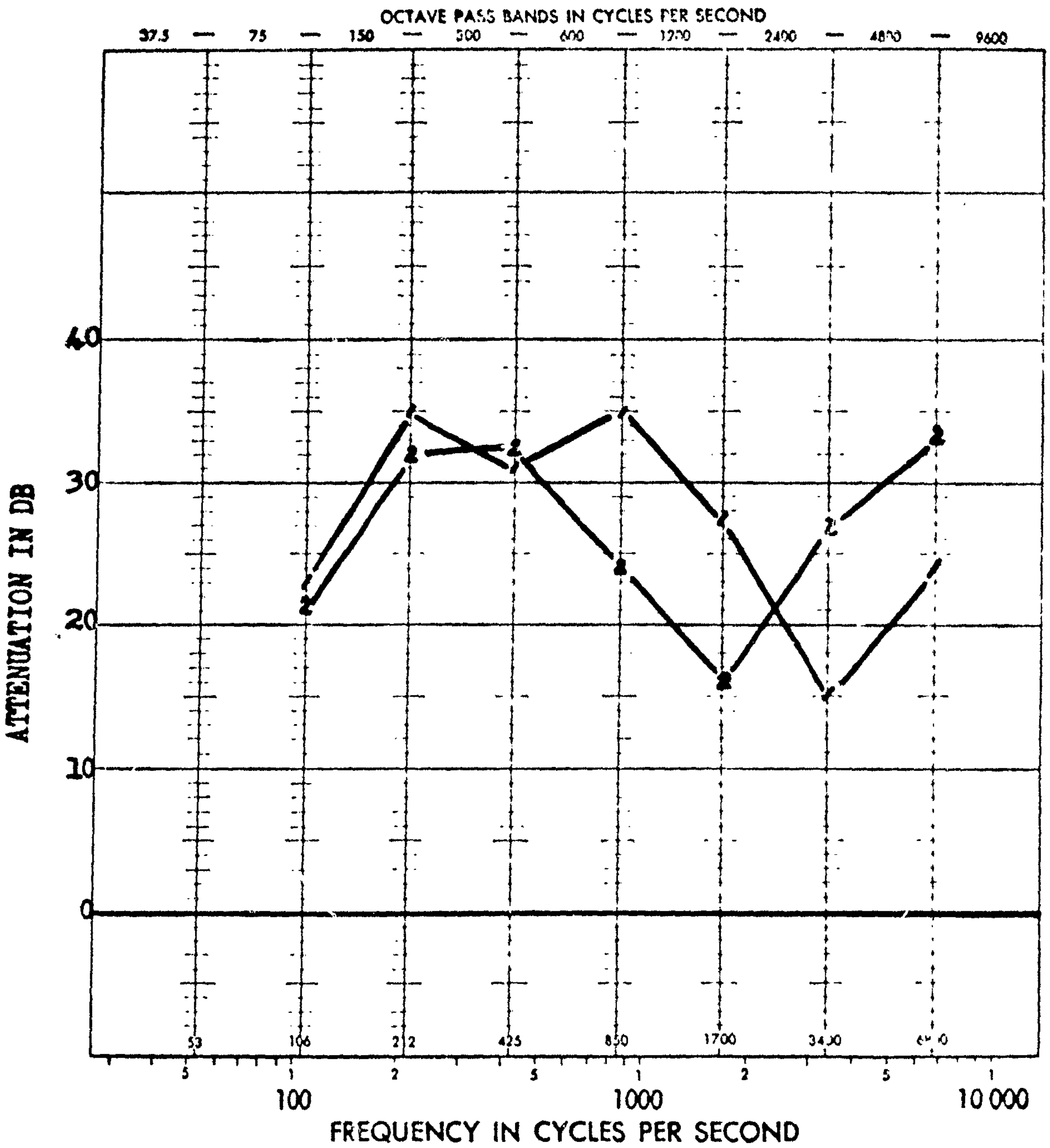


Curve 1: Glass sphere 12" in diameter, flanged opening sealed to 3/4" plywood base.

Curve 2: Truncated glass sphere, sealed to 3/4" plywood base.

Figure A6-2

ATTENUATION OF STEEL AND EXPERIMENTAL PLASTIC HELMETS

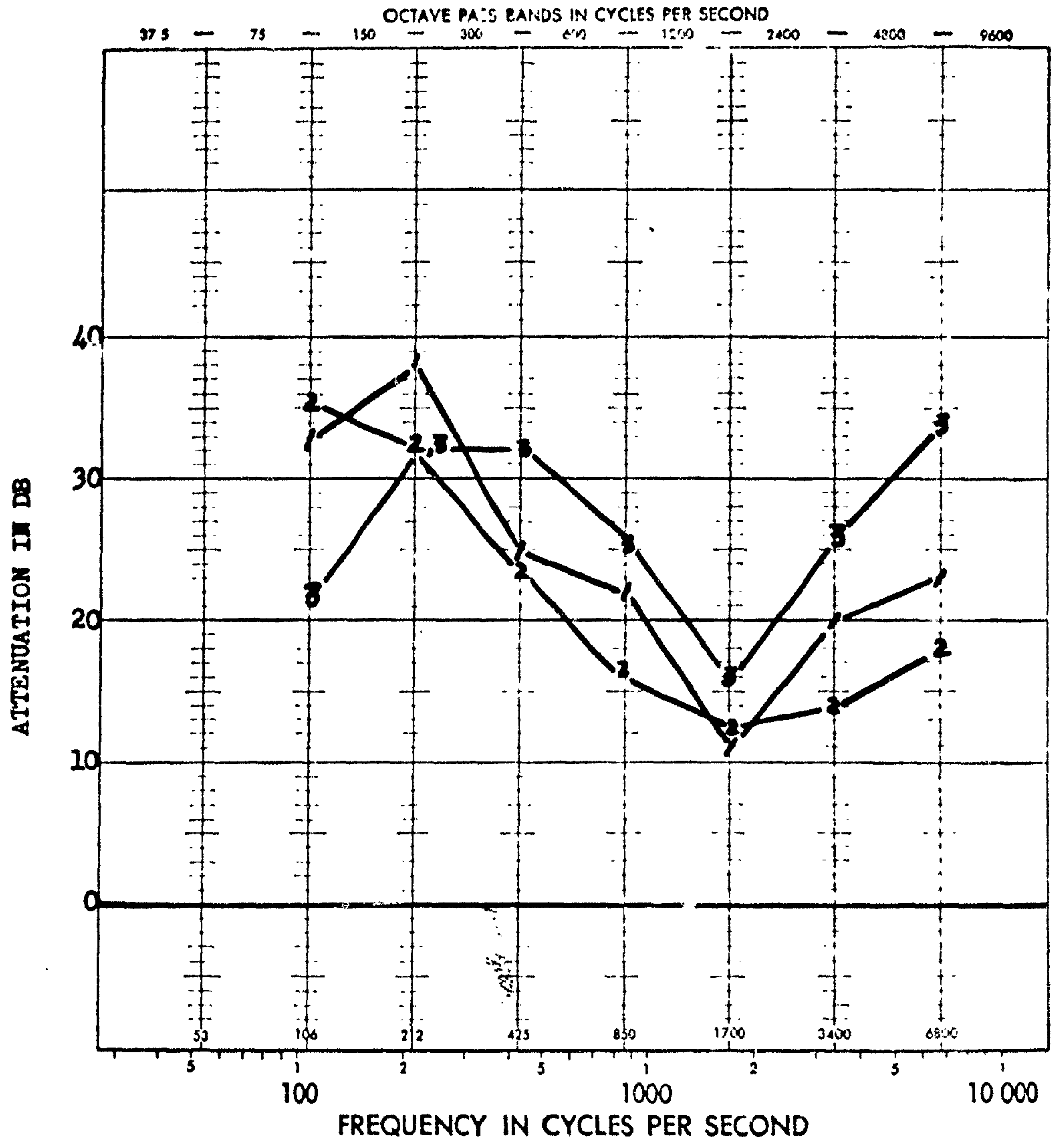


Curve 1: Steel helmet, no absorption, strapped tightly to wooden form.

Curve 2: Experimental plastic helmet, no absorption, strapped tightly to wooden form.

Figure A6-3

ATTENUATION OF EXPERIMENTAL PLASTIC HELMET
DEPENDANCE ON NATURE OF SEAL



Curve 1: Water seal. Average of curves 1-3 Figure A6-5.

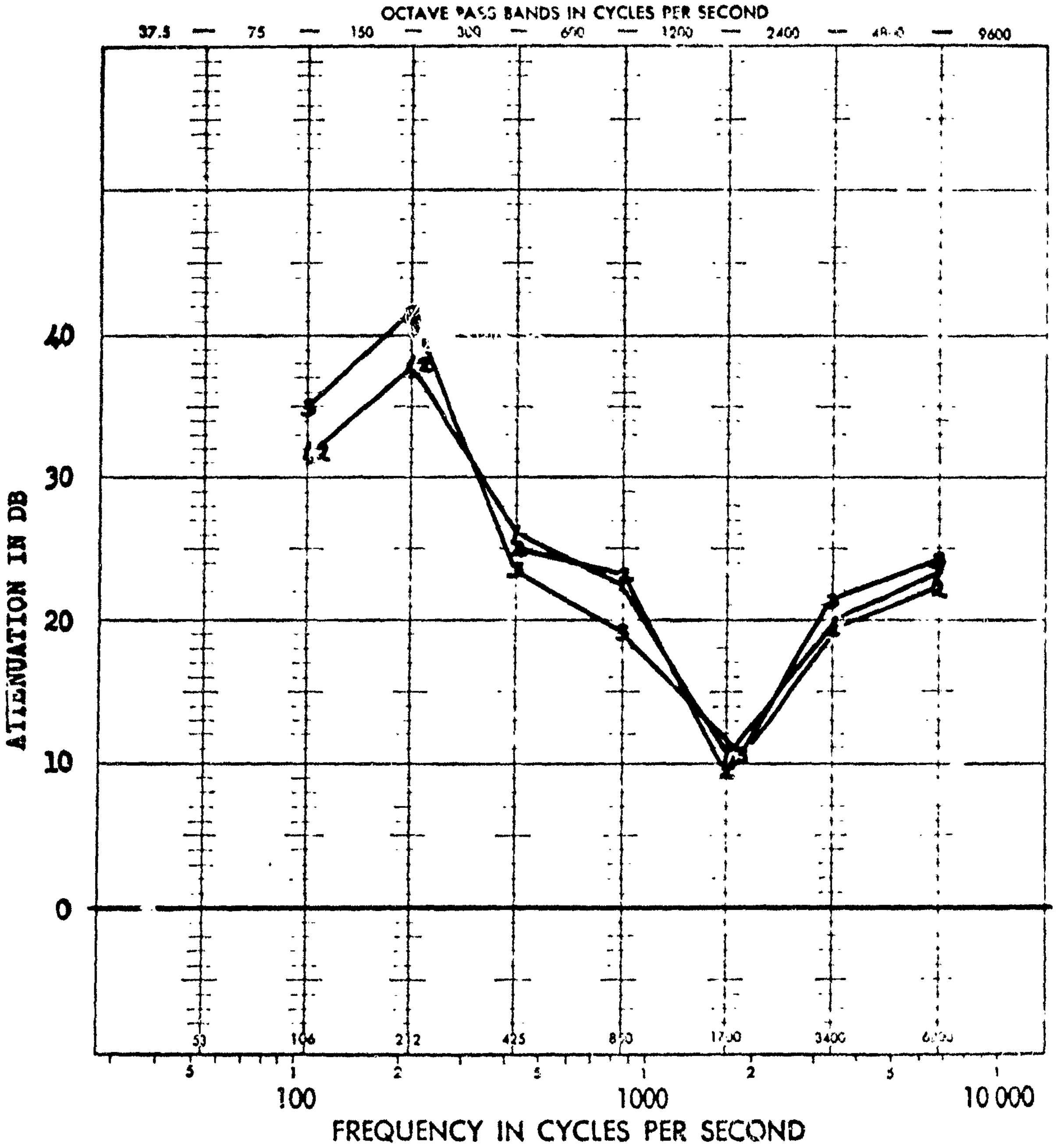
Curve 2: Mortite putty seal - helmet strapped to wooden form.

Curve 3: Foam rubber seal - helmet strapped to wooden form.

Note: No absorption in helmet.

Figure A6-4

ATTENUATION OF EXPERIMENTAL PLASTIC HELMET WITH LIQUID SEAL.



Curve 1: Water seal weight 15 lb. 7 oz. Width/thickness 2.25.

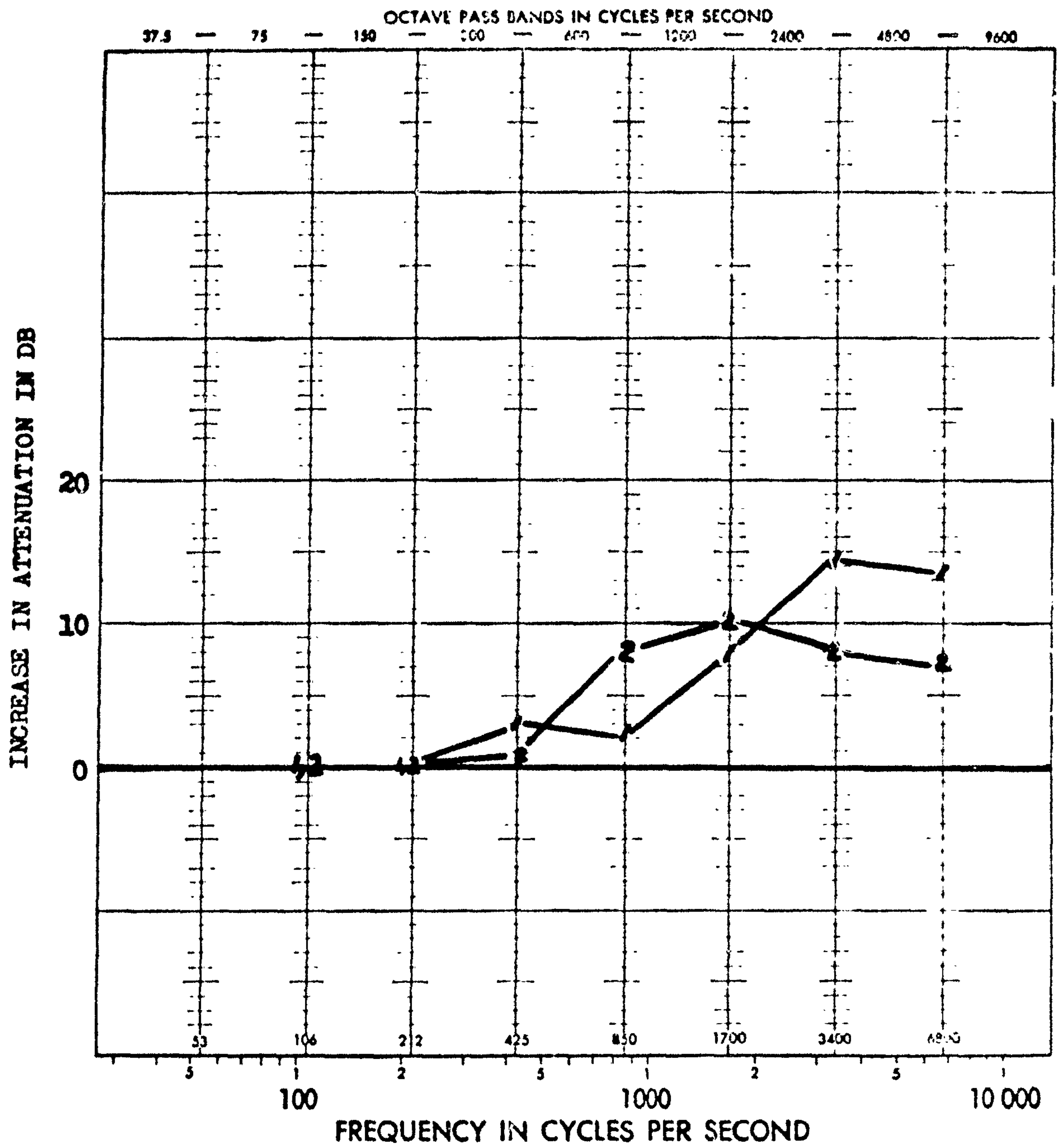
Curve 2: Water seal weight 12 lb. 11 oz. Width/thickness 3.

Curve 3: Water seal weight 7 lb. 13½ oz. Width/thickness 6.

Note: No absorption inside helmet.

Figure A6-5

INCREASE IN ATTENUATION WHEN ABSORBING MATERIAL IS PLACED INSIDE HELMETS



Curve 1: Steel helmet, strapped tightly to wooden form.

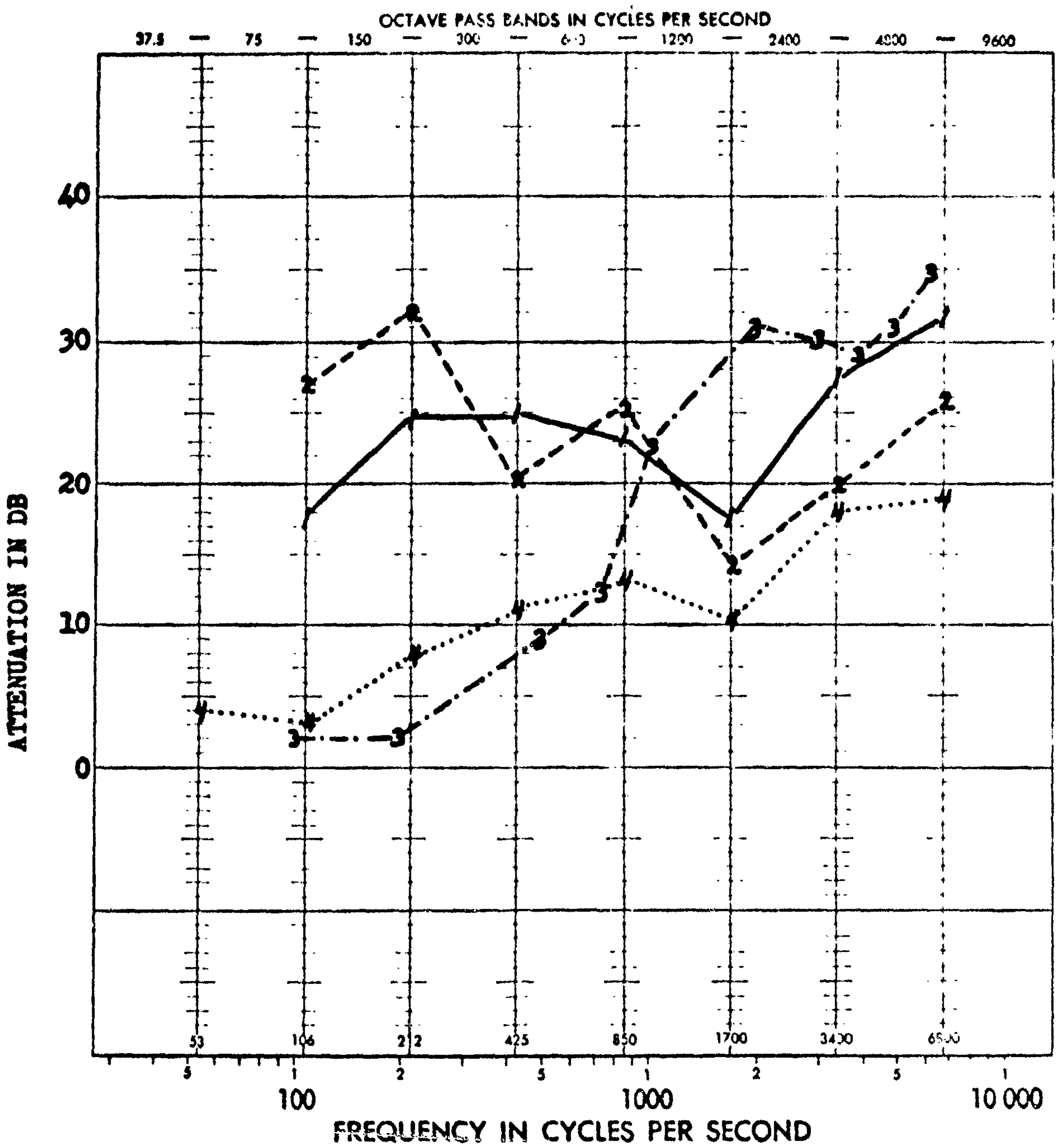
Curve 2: Experimental plastic helmet, strapped tightly to wooden form.

Figure A6-6

EXPERIMENTAL SITUATION FOR THE MEASUREMENT
OF THRESH HOLD OF INTELLIGIBILITY

Fig. A6-7

ATTENUATION OF EXPERIMENTAL AND MA-1 HELMETS ON HUMAN SUBJECTS.



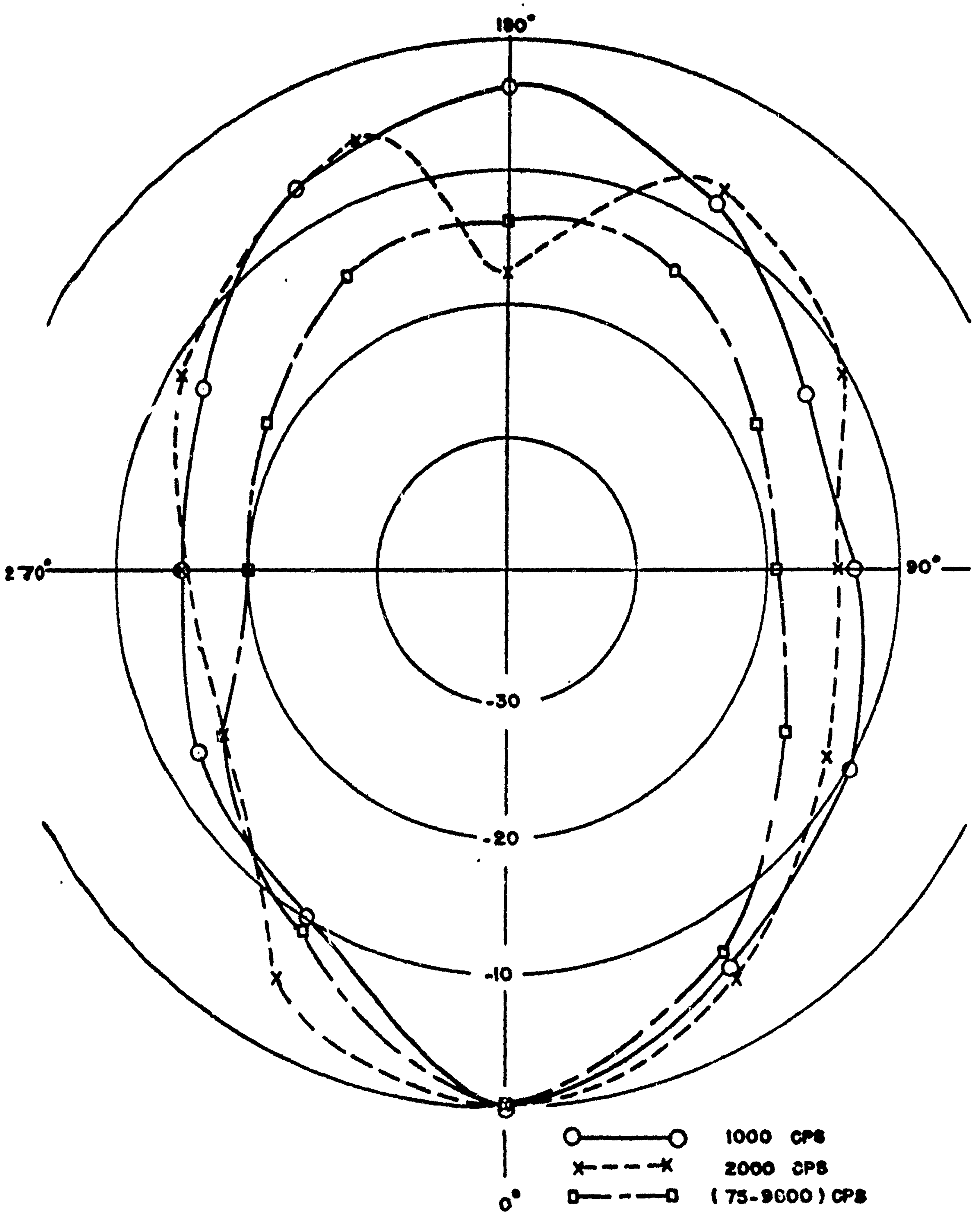
Curve 1: Experimental plastic helmet with 2" foam rubber pad belted tightly to subject. (Measured at ear position.)

Curve 2: Experimental plastic helmet with water ring seal. (Measured at ear position.)

Curve 3: MA-1 helmet (measured at ear position) RAC Reports, Contract AF33(616)456.

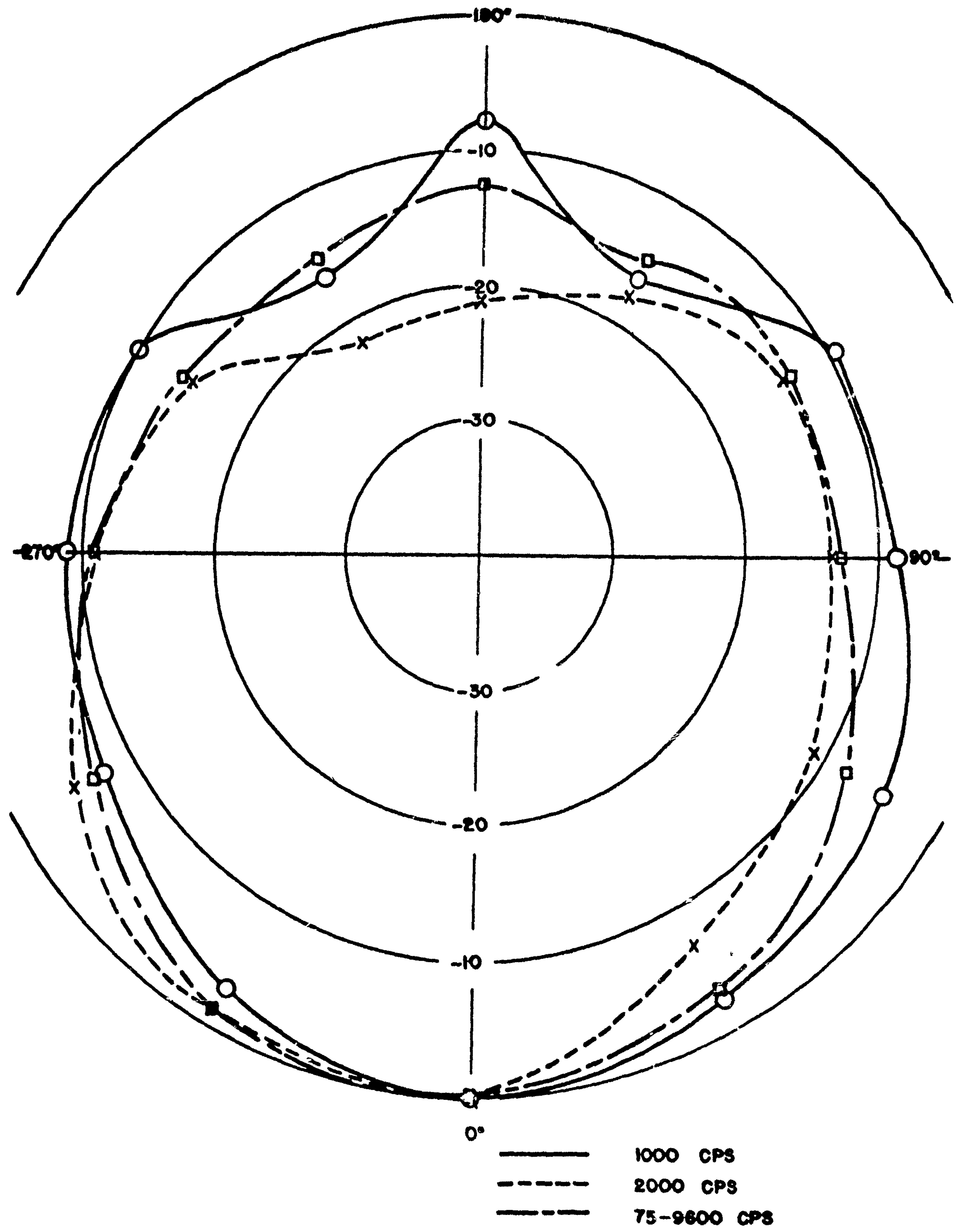
Curve 4: MA-1 helmet (measured at lip position) WEAL data.

Figure A6-8



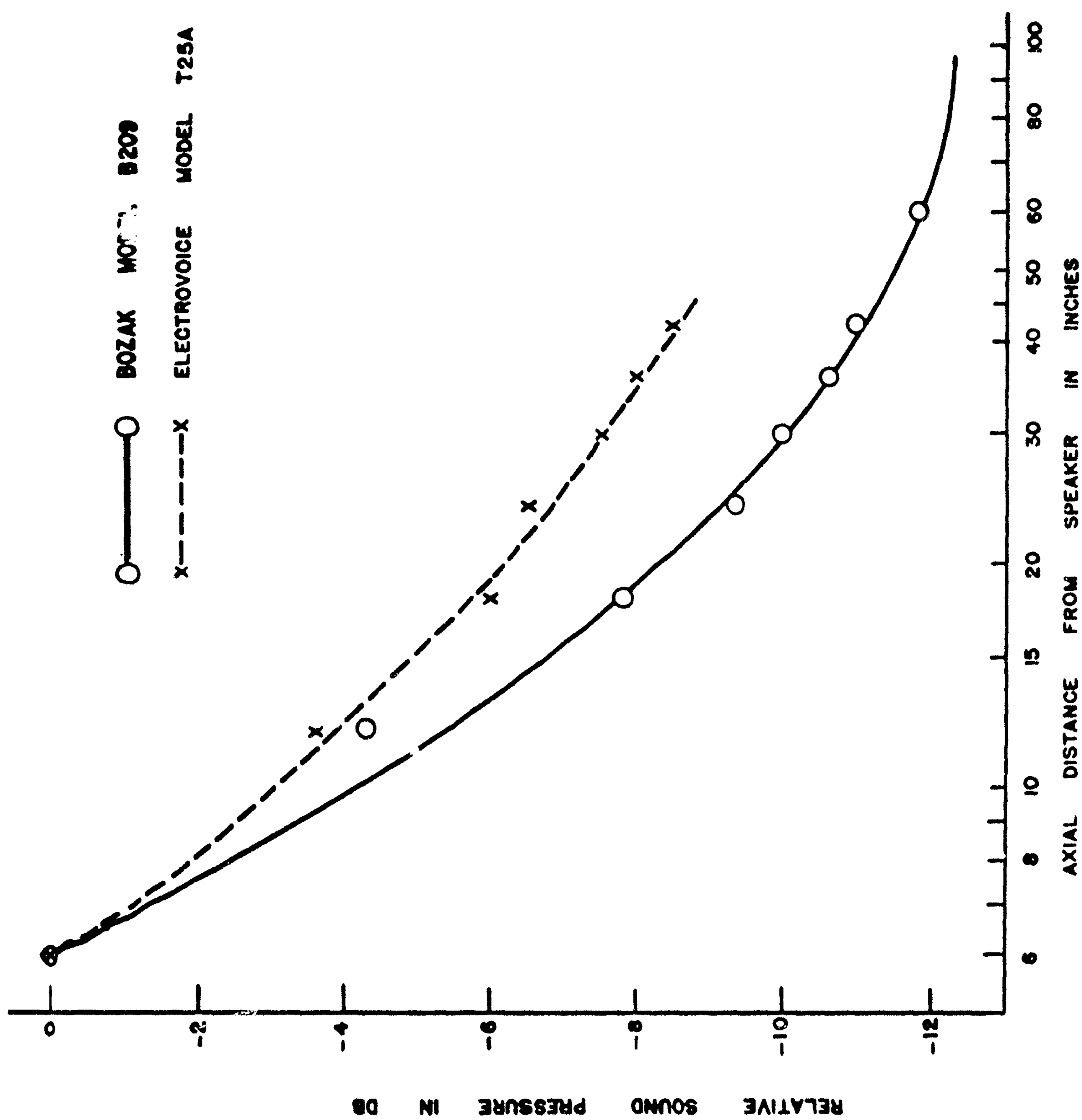
- POLAR RESPONSE OF ELECTROVOICE MODEL T25A SPEAKER

Figure A6-9



- POLAR RESPONSE OF BOZAK MODEL B209 SP_AKER

Figure A6-10



- RELATIVE AXIAL SOUND PRESSURE LEVEL OF SPEECH PROJECTION SPEAKERS IN REVERBERATION ROOM

Figure A6-11

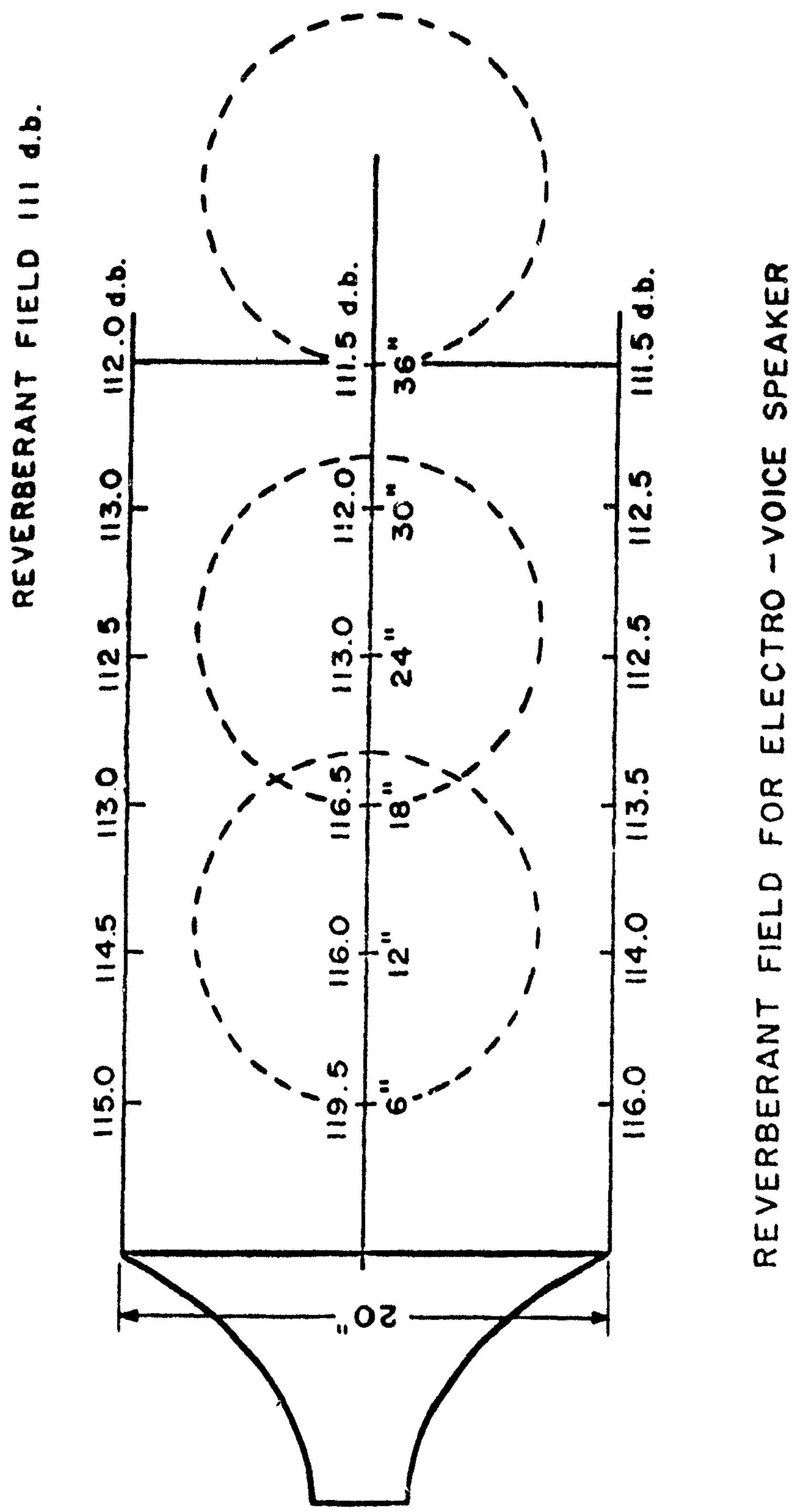
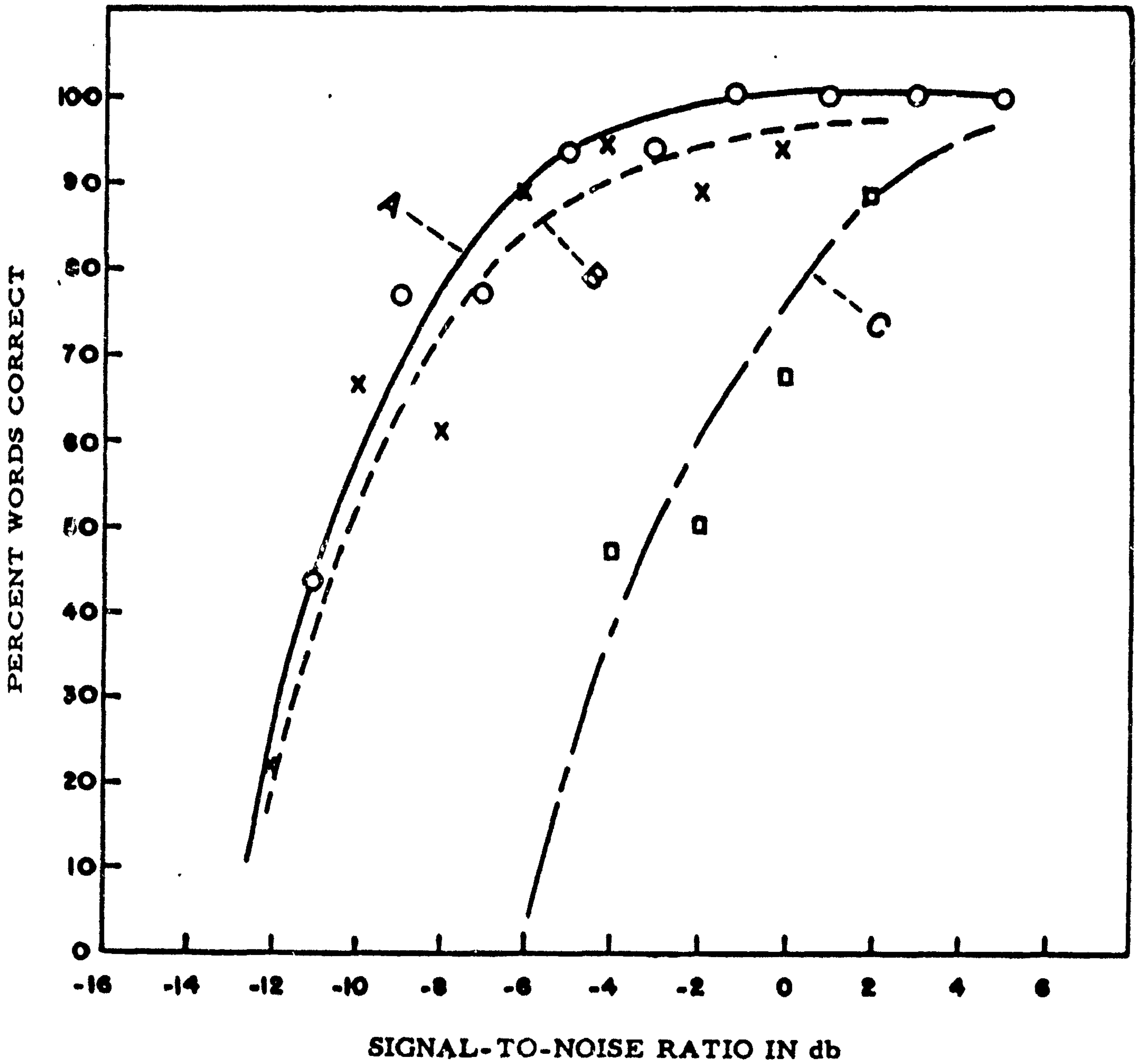
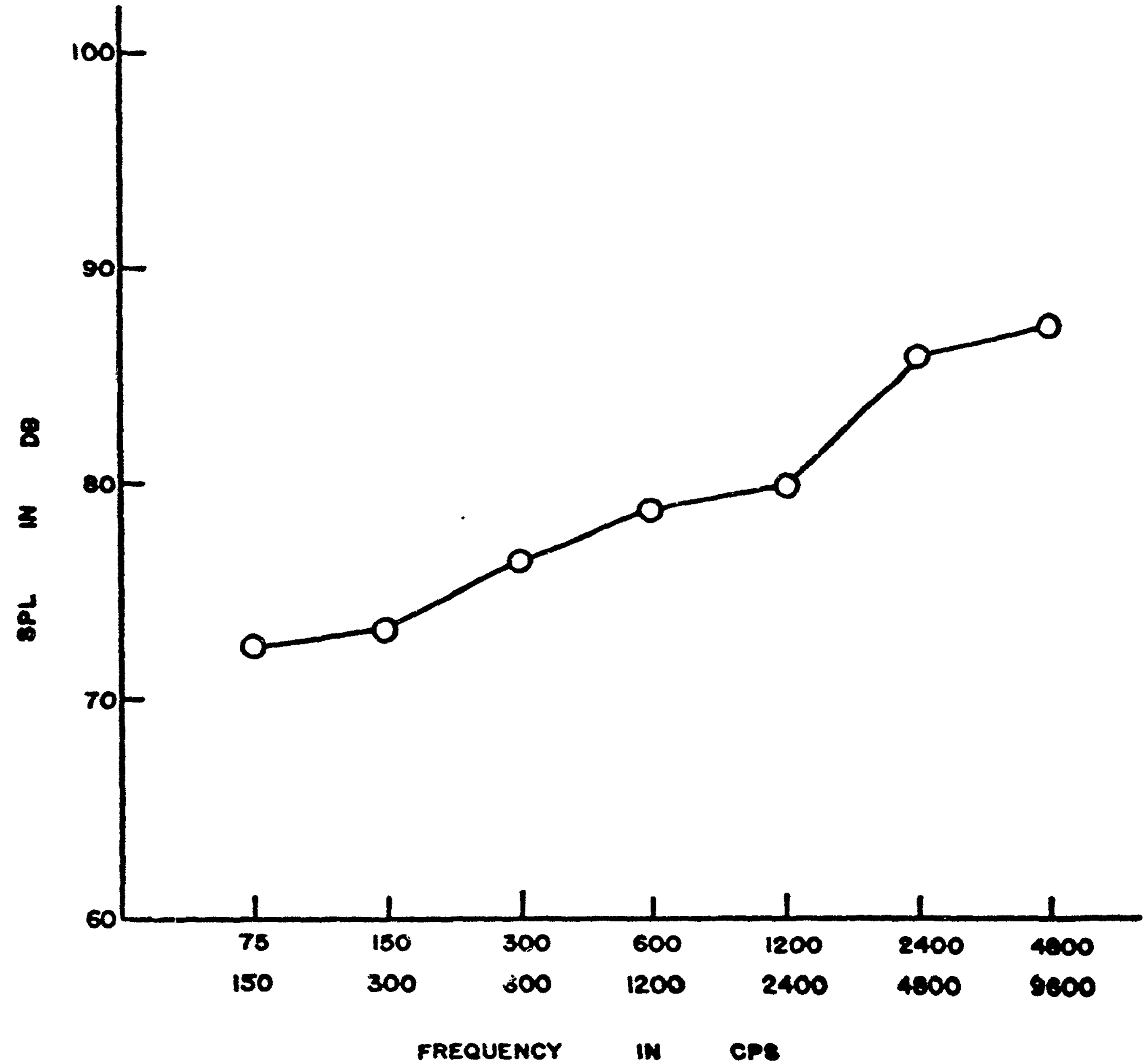

REVERBERANT FIELD 111 d.b.
115.0
114.5
113.0
112.5
113.0
112.0 d.b.
119.5
116.0
116.5
113.0
112.0
111.5 d.b.
6"
12"
18"
24"
30"
36"
116.0
114.0
113.5
112.5
112.5
111.5 d.b.
20"
REVERBERANT FIELD FOR ELECTRO-VOICE SPEAKER

Figure A6-12



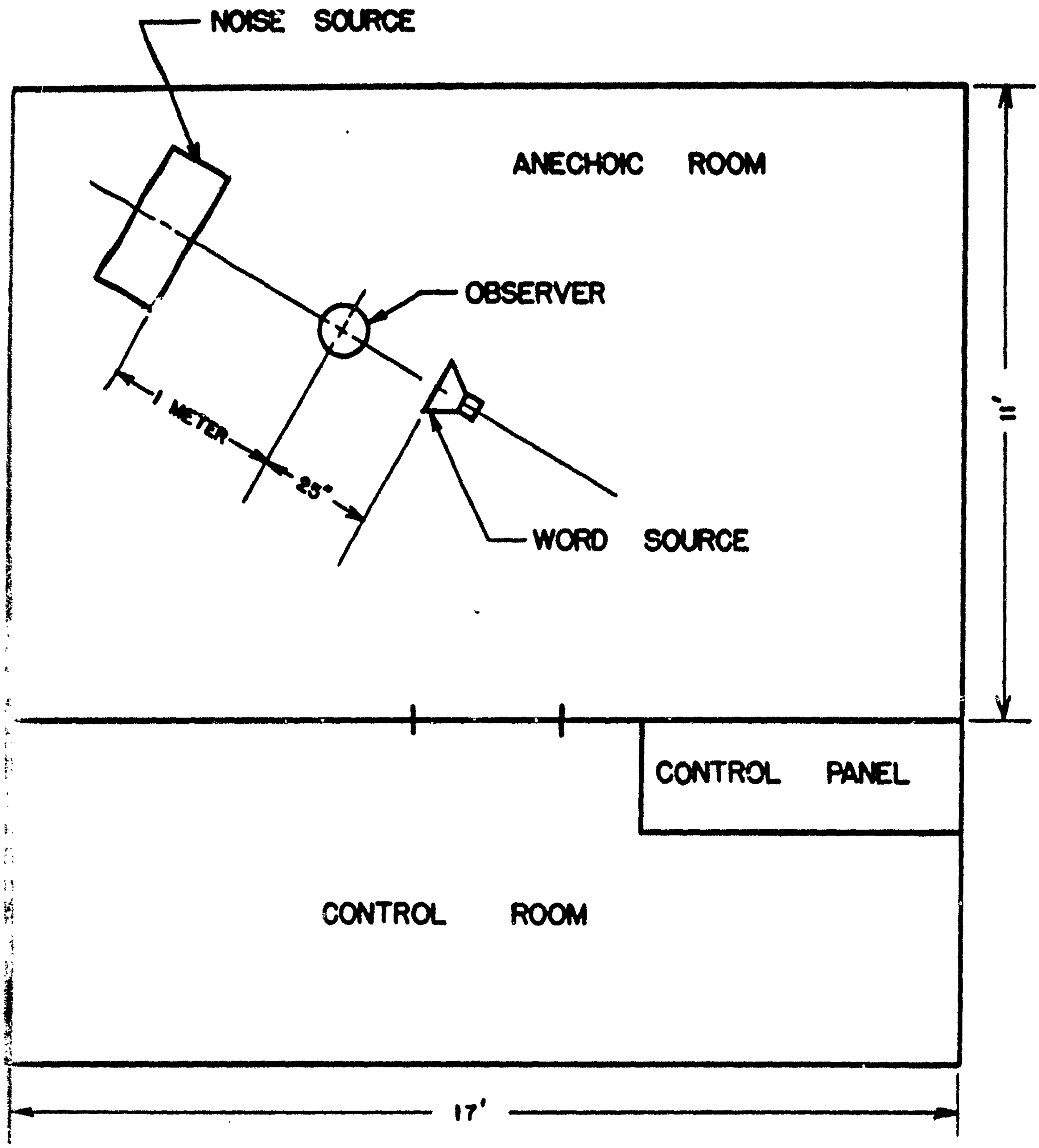
- Intelligibility of Spondee Words in the Experimental Helmet as a Function of Signal-to-Noise Ratio at the Position of the Listener. Distances from the Electro-voice Speaker to the Helmet were: Curve A: 6"; Curve B: 18"; Curve C: 36". Noise SPL = 110 db re .0002 Microbar.

Figure A6-13



- SPECTRUM OF WHITE NOISE USED IN TESTING FOR INTELLIGIBILITY OF PB WORDS IN AN ANECHOIC ENVIRONMENT. OVERALL LEVEL 90 DB.

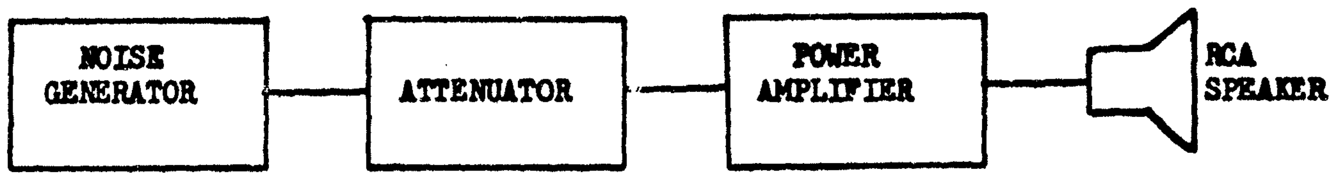
Figure A6-14



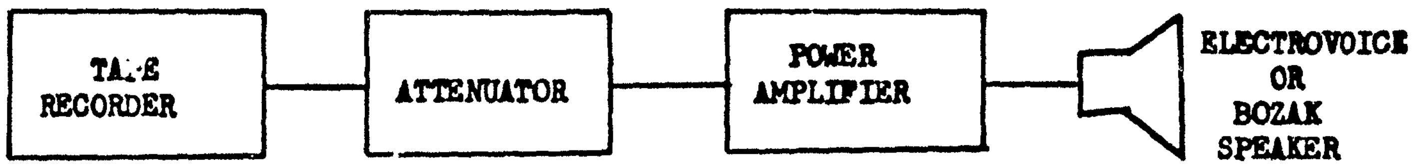
- PLAN VIEW OF ANECHOIC ROOM SHOWING ORIENTATION OF SPEAKERS AND LISTENER FOR PB WORD TESTS

Figure A6-15

NOISE GENERATING SYSTEM

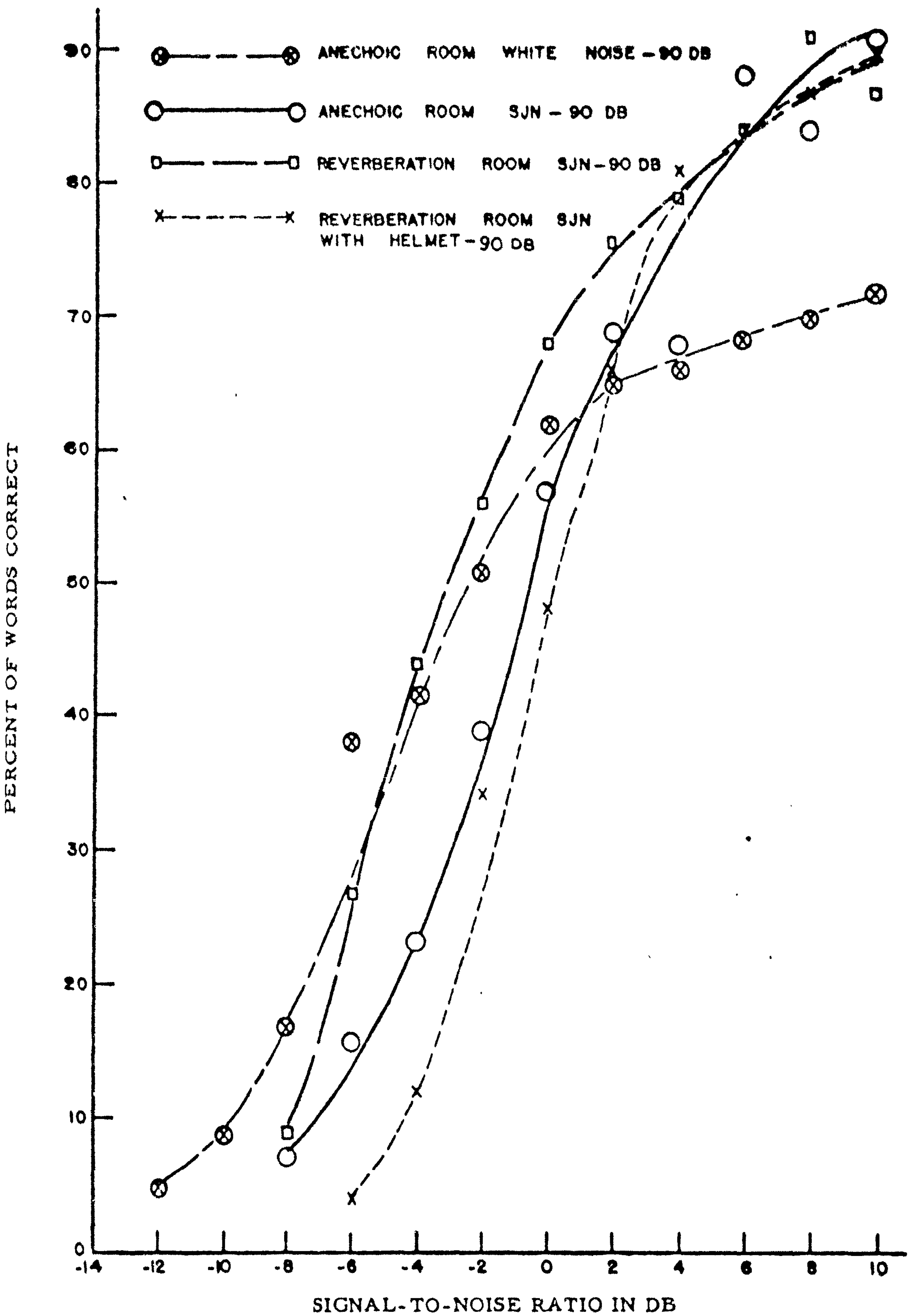


SPEECH TRANSMISSION SYSTEM



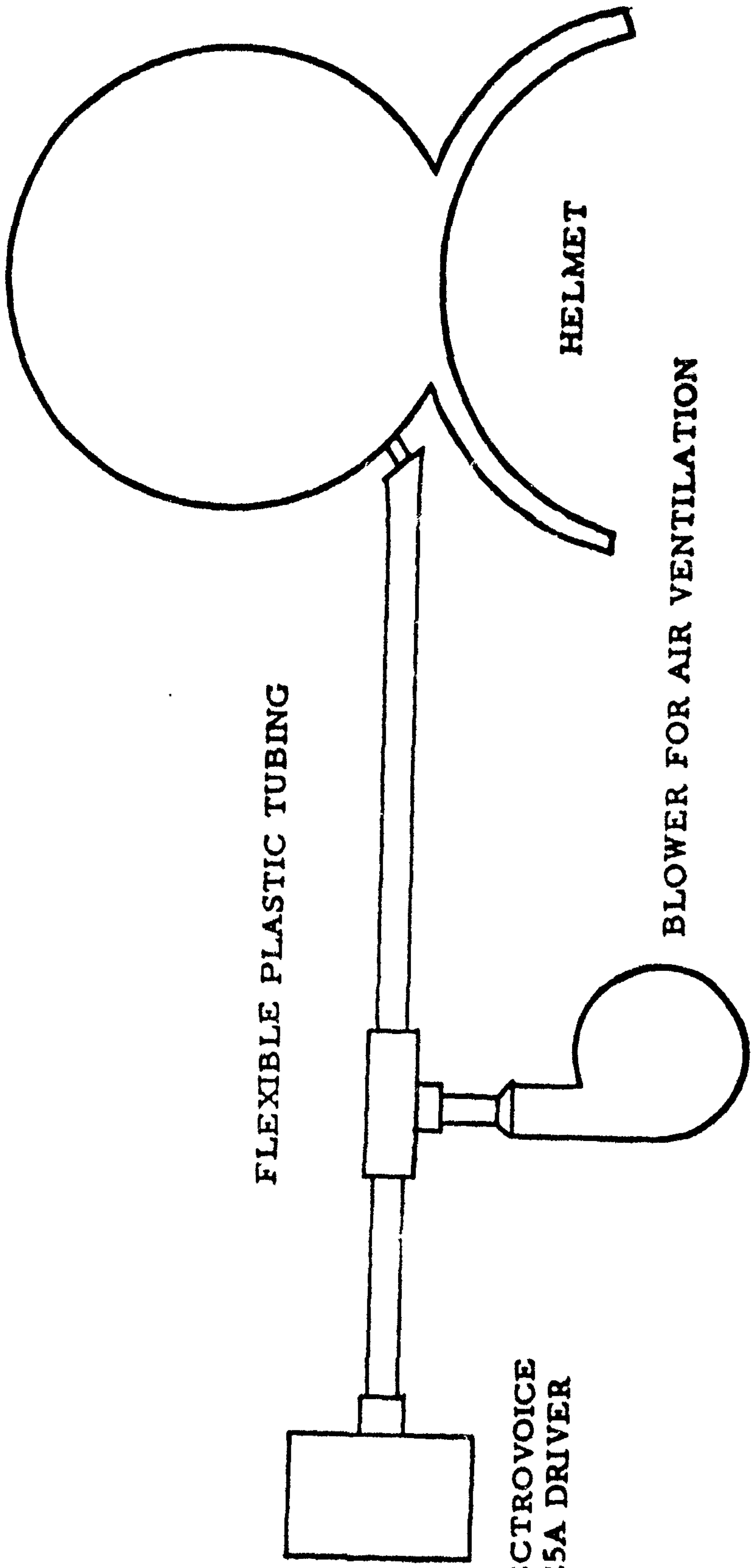
BLOCK DIAGRAMS OF CIRCUITS USED IN TESTING

Figure A6-16



- ARTICULATION SCORES FOR PB WORDS

Figure A6-17



- DETAILS OF EXPERIMENTS ON TRANSMISSION OF SOUND THROUGH TUBES

Figure A6-18

INSTRUMENTATION FOR CLIPPED SPEECH

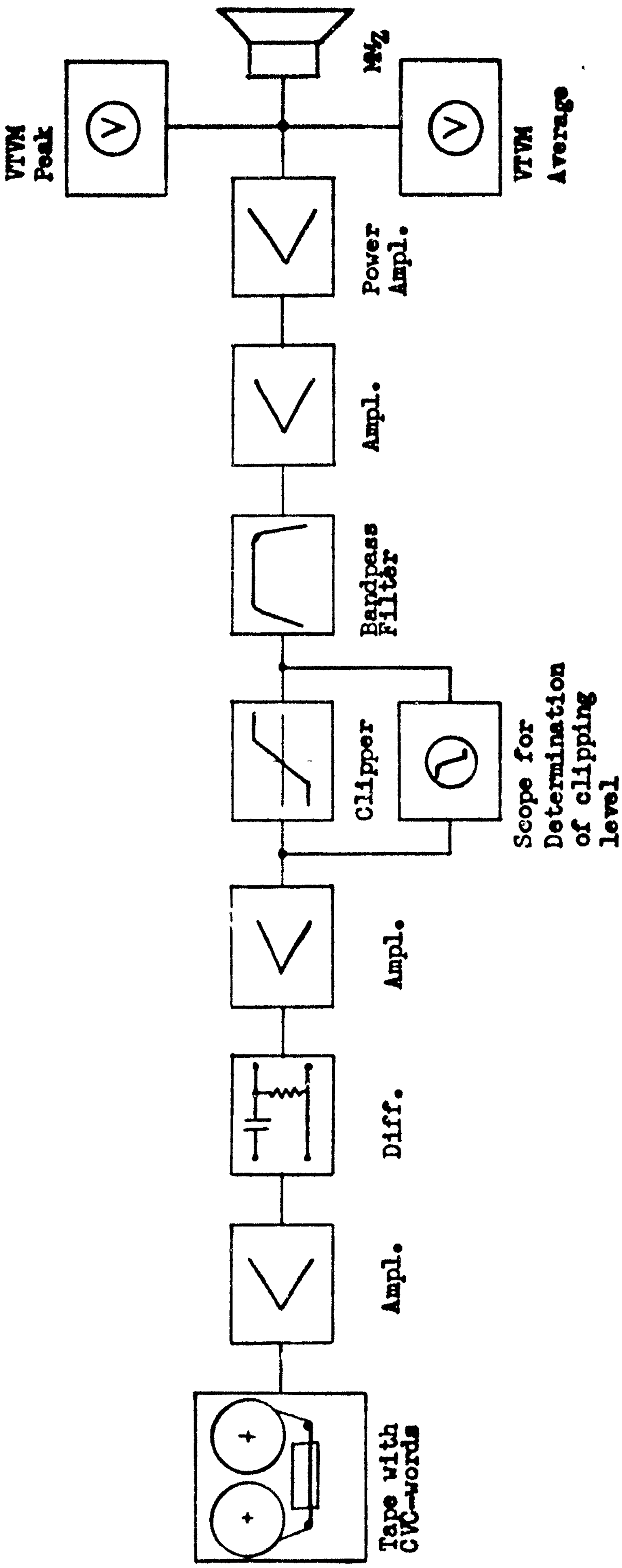


Figure A6-19

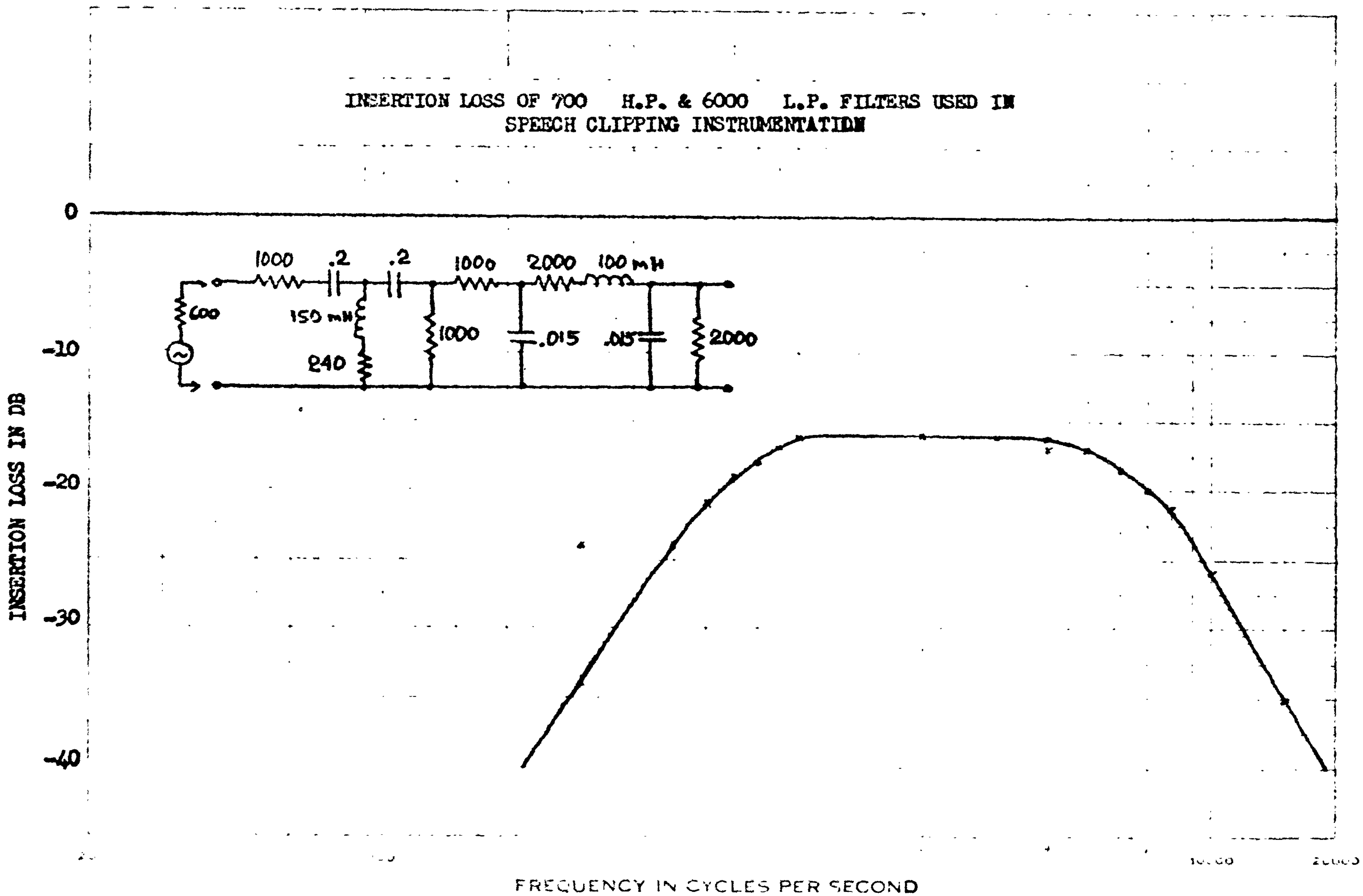
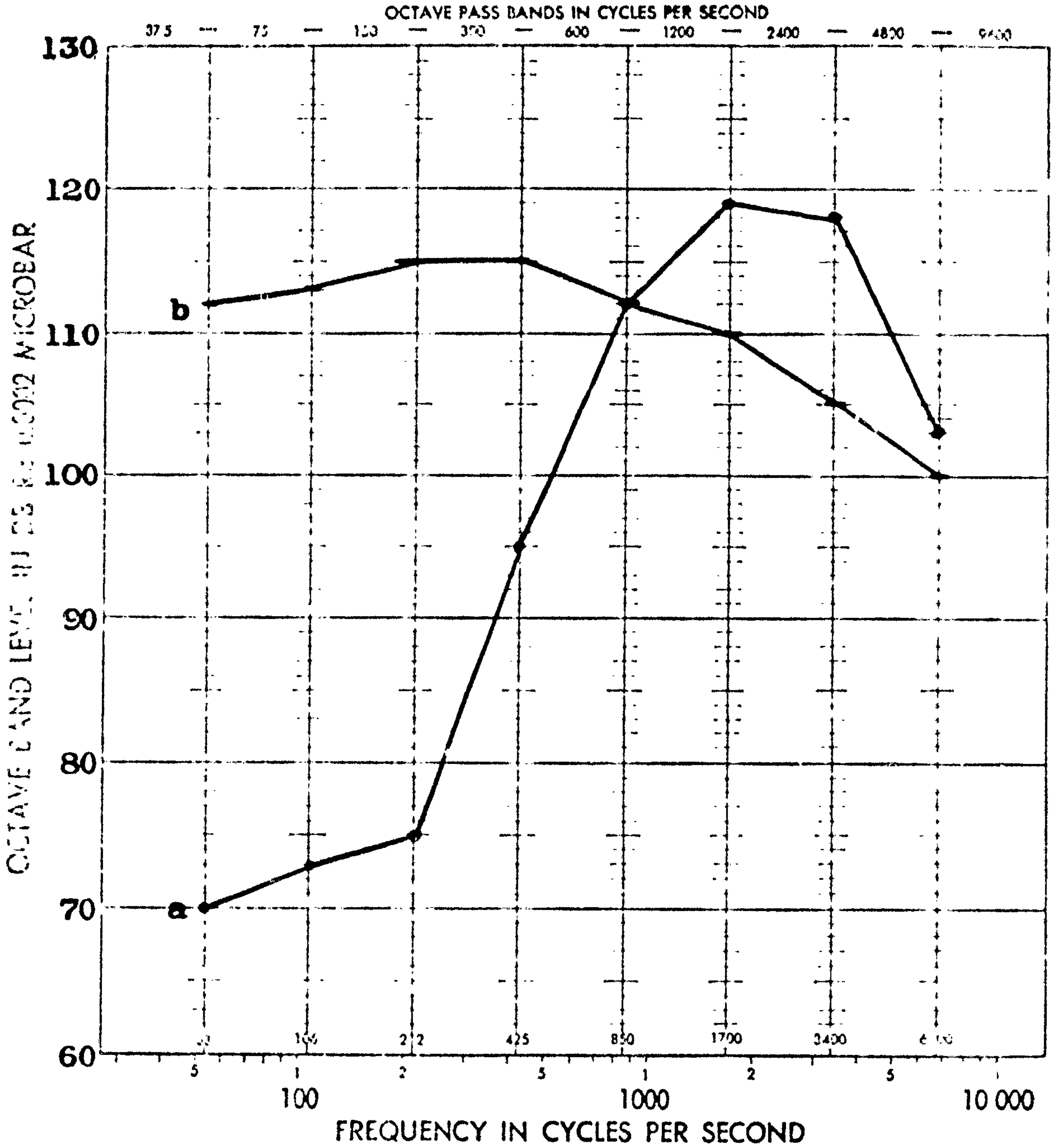

INSERTION LOSS OF 700 H.P. & 6000 L.P. FILTERS USED IN
SPEECH CLIPPING INSTRUMENTATION
INSERTION LOSS IN DB
FREQUENCY IN CYCLES PER SECOND
0
-10
-20
-30
-40
1000
.2
.2
1000
2000
100 mH
600
150 mH
240
1000
.015
.015
2000

Figure A6-20

LEVEL OF CLIPPED SPEECH IN NOISE ENCLOSURE

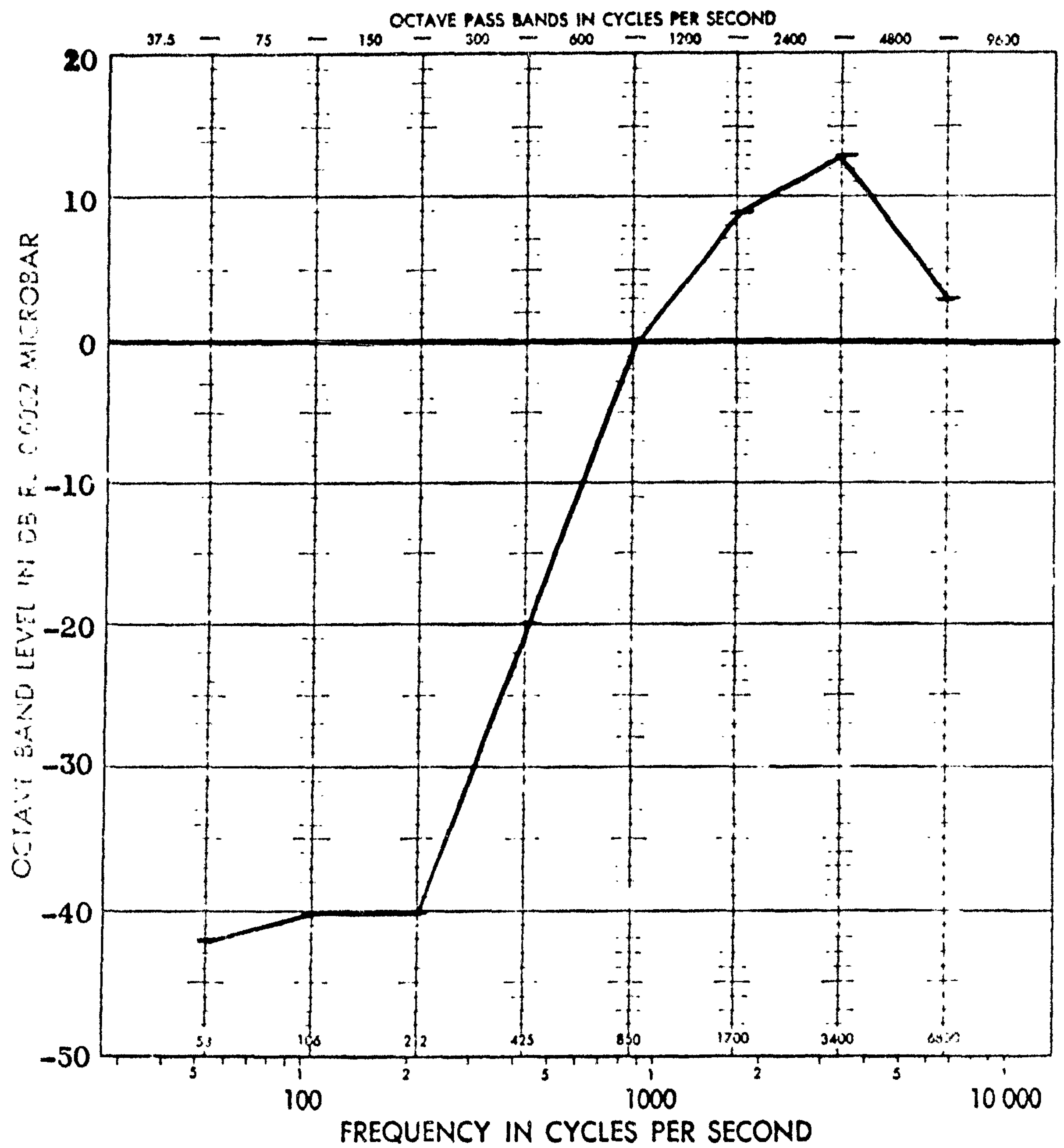


Curve a : Level of clipped , filtered speech in the noise enclosure. (Overall level: 124 db).

Curve b : Level of jet noise in noise enclosure. (Overall level: 121 db)

Figure A6-21

SIGNAL/NOISE RATIO FOR CLIPPED SPEECH IN JET NOISE.



Overall signal-to-noise ratio: + 3 db.

Figure A6-22

Table A6 - I

Electrical power in watts required to produce a SPL of 100 db re .0002 microbar.

D in inches	Electrovoice Model T25A (db re 1 watt)	Bozak Model B209 (db re 1 watt)
6	-14.5	-8.2
12	-11.3	-3.2
18	-8.3	-0.7
24	-8.0	+1
30	-7.0	+1.8
36	-6.5	+2.3
42	-6.0 *	+2.8
60		+3.6 *

D = Distance from mouth of speaker along axis.

* = At this distance we are essentially in the reverberant field of the speaker.

INTELLIGIBILITY OF SPONDEE WORDS IN SYNTHETIC JET NOISE
OBSERVER WEARING HELMET

Electrovoice, Horn Type Speaker, Model T25A

Sound Pressure Level of Synthetic Jet Noise in db re .0002 Microbar	D	S/N Ratio in db for 50 percent Articulation Score	
		CC	VR
120	13		-8
	25	- 4	-5
	43	- 4	
110	13	- 9	-5
	25	-10	-7
	43	- 4	0
100	13	-10	-8
	25	- 9	-8
	43	- 6	-4

D represents distance in inches from mouth of speaker to position occupied by observer's head during test.

Table A6-II

INTELLIGIBILITY OF SPONDEE WORDS IN SYNTHETIC JET NOISE
OBSERVER WEARING HELMET

Bozak Speaker, Model B209
In one cu ft closed box baffle

Sound Pressure Level of Synthetic Jet Noise in db re .0002 Microbar	D	S/N Ratio in db for 50 percent Articulation Score	
		CC	VR
110	13	- 5	-8
	25	-10	-9
	43	- 4	
100	13	- 6	-5
	25	-10	-7
	43	- 3	

D represents the distance in inches from mouth of speaker to position occupied by observer's head during testing.

Table A6-III

INTELLIGIBILITY OF SPONDEE WORDS IN SYNTHETIC JET NOISE OBSERVER NOT WEARING HELMET

Electrovoice Speaker, Model T25A

Sound Pressure Level of Synthetic Jet Noise in db re .0002 Microbar	D	S/N Ratio in db for 50 percent Articulation Score	
		CC	VR
100	6	-12	- 7
	18	-10	-10
	36	-10	- 7
90	6	- 9	- 9
	18	-10	- 9
	36	-12	-10
80	6	-12	-11
	18	-12	-11
	36	-13	-10

D represents distance in inches from mouth of speaker to position of observer's head during testing.

Table A6-IV

RESULTS OF INTELLIGIBILITY TESTING WITH PB WORDS

(Percent of Words Correctly Identified)

a) Listening in anechoic room with masking by white noise at SPL of 90 db

S/N	+10	8	6	4	2	0	-2	-4	-6	-8	-10	-12	-14	-16	-18	-20	-22
VR	59	66	64	74	56	60	51	34	28	12	0						
CC	66	58	53	46	58	51	32	34	37	30	26	14					
DL	92	86	88	78	80	74	71	57	48	10	0						
Avg.	72	70	68	66	65	62	51	42	38	17	9	5					
LS	92	82	84	88	88	84	82	88	80	82	70	64	56	44	26	28	10

b) Listening in anechoic room with masking by jet noise at SPL of 90 db

S/N	+10	8	6	4	2	0	-2	-4	-6	-8	-10	-12	-14	-16	-18
VR	88	79	84	68	54	52	14	0	0	0					
CC	90	88	88	68	70	50	46	28	22	22					
DL	94		91		82	69	57	4[illegible]	16	0					
Avg.	91	84	88	68	69	57	39	23	16	7					
LS	98	94	88	95	90	96	90	90	90	90	84	61	44	38	6

c) Listening in reverberation room with masking by jet noise at SPL of 90 db

S/N	+10	8	6	4	2	0	-2	-4	-6	-8
VR	87	90	86	78	74	74	62	55	42	10
CC	82	86	75	72	67	52	41	28	10	0
DL	9[illegible]	96	90	88	87	77	66	50	28	16
Avg.	87	91	84	79	76	68	56	44	27	9
LS	92		94	90	92	89	80	55	32	8

d) Listening in reverberation room with masking by jet noise at SPL of 90 db. Observer wearing plastic helmet.

S/N	+10	8	6	4	2	0	-2	-4	-6
VR	98	89	85	87	71	52	45	24	9
CC	82	85	84	75	62	44	22	0	0
Avg.	90	87	84	81	66	48	34	12	4

e) Listening in reverberation room with masking by jet noise at SPL of 105 db. Observer wearing plastic helmet.

S/N	+10	8	6	4	2	0	-2	-4	-6
VR	88	79	78	82	65	56	38	28	0
CC	73	76	66	80	70	54	50	38	30
Avg.	81	78	72	81	68	55	44	33	15
LS	90	74	76	78	60	58	52	24	18
Avg.	84	76	73	80	65	56	47	30	16

TABLE A6-V

COMPARISON OF ELECTRICAL INPUT TO SPEAKER WHEN COMMUNICATING WITH SPEAKER EXTERIOR TO HELMET AND WITH SPEAKER ATTACHED TO AIR SUPPLY LINE. SPONDEE WORDS WERE USED.

A) Using Speaker Exterior to Helmet

SPL of Masking Jet Noise in db re .0002 Microbar	D	S/N Required for 50 percent Intelligibility	Voltage on Speaker db re 1 volt
120	13	-8	20.3
	25	-4	28.6
	43	-4	30.6
110	13	-7	11.6
	25	-9	19.1
	43	-2	21.1

B) Speech Introduced Through Tube

SPL of Masking Jet Noise in db re .0002 Microbar	Length of Tube	Voltage on Speaker db re 1 volt
120	27	0.0
110	71	- 6.0
	45	-12.8
	27	- 9.1

D is distance in inches from mouth of speaker to position occupied by observer's head.

TABLE A6-VI

TABLE A6-VII

SUMMARY OF ELECTRICAL POWER
REQUIRED TO ACHIEVE A 50%
ARTICULATION SCORE

OVER ALL JET NOISE LEVEL (DB RE .0002 BAR)	DISTANCE FROM LOUDSPEAKER (INCHES)	S/N (DB) FOR 50% SCORE	APPROXIMATE POWER REQUIRED (DB RE 1 WATT)
1. Spondee words. Subject in reverberation room. No helmet. Electrovoice speaker, Model T 25A			
100	6	-10	
	18	-10	
	36	-8	
90	6	-9	
	18	-10	
	36	-11	
2. Spondee words. Subject in reverberation room. Plastic helmet with foam rubber seal. Electromoice speaker, model T 25A			
120	13	-8	+0.7
	25	-5	+7.0
	43	-4	+10.0
110	13	-7	-8.3
	25	-8	-6.0
	43	-2	+2.0
100	13	-9	-20.3
	25	-9	-17.0
	43	-5	-11.0
3. Spondee words. Subject in reverberation room. Plastic helmet with foam rubber seal. Bozak Speaker, Model B209			
110	13	-6	+1.2
	25	-10	+1.0
	43	-4	+8.8
100	13	-6	-9.2
	25	-8	-7.0
	43	-3	+0.2
4. Spondee words. Subject in reverberation room. Communication via air supply tubing. Helmet with foam rubber seal. E V speaker, Model T 25A.			
120	27*		
110	71		
	45		
	27		

* Length of Tube

TABLE A6-VII

TABLE A6 - VIII

SUMMARY OF ELECTRICAL POWER REQUIRED TO ACHIEVE A 50% ARTICULATION SCORE

OVERALL JET NOISE LEVEL (DB RE .0002 BAR)	DISTANCE FROM LOUDSPEAKER (INCHES)	S/N RATIO (DB) FOR 50% SCORE	APPROXIMATE POWER REQUIRED (DB RE 1 NEW)
1. PB words. Subject in reverberation room. No helmet. EV speaker, Model T25 A.			
90	25	-3	-21
2. PB words. Subject in reverberation room. Plastic helmet, foam seal. EV speaker, Model T25A.			
90	25	+1	-17
105	25	-1	-4

TABLE A6-VIII